DEFENSE INFORMATION SYSTEMS AGENCY
P. O. BOX 4502
ARLINGTON, VIRGINIA 22204-4502

IN REPLY REFER TO: Joint Interoperability Test Command (JTE)

MEMORANDUM FOR DISTRIBUTION **29 Dec 10**

SUBJECT: Special Interoperability Test Certification of the Avaya Aura™ AS5300 Local Session Controller, Version 2.0 (with specified patch releases)

References: (a) DoD Directive 4630.05, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01E, "Interoperability and Supportability of Information Technology and National Security Systems," 15 December 2008
(c) through (e), see Enclosure 1

1. References (a) and (b) establish the Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Avaya Aura™ AS5300, Version 2.0 (with specified patch releases), hereinafter referred to as the System Under Test (SUT) is certified for joint use in the Defense Information System Network (DISN) as a Local Session Controller (LSC). The fielding of the SUT is limited to IP version 4 (IPv4) across the DISN based on the fielding environment and a Plan of Action and Milestones (PoAM) addressing critical IP version 6 (IPv6) discrepancies by 30 April 2011. Intra-enclave use of IPv4 and IPv6 is authorized for use. The certification status of the SUT will be verified during operational deployment. Any new discrepancy noted in the operational environment will be evaluated for impact on the existing certification. These discrepancies will be adjudicated to the satisfaction of DISA via a vendor PoAM, which will address all new critical TDRs within 120 days of identification. Testing was conducted using LSC product requirements derived from, Reference (c), and LSC test procedures derived from Reference (d). No other configurations, features, or functions, except those cited within this memorandum, are certified by JITC. This certification expires upon changes that affect interoperability, but no later than three years from the date the SUT was posted on the Unified Capabilities (UC) Approved Products List (APL) (1 September 2010).

3. This finding is based on interoperability testing conducted by JITC, review of the vendor's Letters of Compliance (LoC), and DISA Information Assurance (IA) Certification Authority (CA) approval of the IA configuration. Interoperability testing was conducted by JITC, Fort Huachuca, Arizona, from 12 October through 30 November 2009. Review of the vendor's LoC was completed on 21 September 2010. The DISA CA has reviewed the IA Assessment Report for the SUT, Reference (e), and based on the findings in the report has provided a positive recommendation. The acquiring agency or site will be responsible for the DoD Information Assurance Certification and Accreditation Process (DIACAP) accreditation. The JITC certifies

the SUT. Enclosure 2 documents the test results and describes the tested network and system configurations including specified patch releases.

4. The interface, Capability Requirements (CR) and Functional Requirements (FR), and component status of the SUT is listed in Tables 1 and 2. The threshold Capability/Functional requirements for LSCs are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of Reference (c) and were used to evaluate the interoperability of the SUT.

**Table 1. SUT Interface Interoperability Status**

| Interface | Critical | UCR Reference | Threshold CR/FR Requirements (See note 1.) | Status | Remarks (See note 2.) |
|---|---|---|---|---|---|
| **Line Interfaces** | | | | | |
| 10Base-X | Yes | 5.3.2.6.3 | 2, 4, 10, 13, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to PEIs (voice) and Softphones (voice and video). |
| 100Base-X | Yes | 5.3.2.6.3 | 2, 4, 10, 13, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3u. Applies to PEIs (voice) and Softphones (voice and video). |
| 1000Base-X | No | 5.3.2.6.3 | 2, 4, 10,13, 16 | Not Tested | This interface is not offered by the SUT PEIs. |
| 2-wire analog | Yes | 5.3.2.6.1.6 | 2, 4, 10, 13, | Certified | Met threshold CRs/FRs for 2-wire instruments. Applies to 2-wire secure and non-secure analog instruments. Requirement met through use of an IAD that supports IEEE 802.3i, 802.3u, and 802.3ab. |
| BRI | No | 5.3.2.6.1.8 | 2, 4, 10, 13 | Not Tested | This interface is not supported by the SUT. |
| **External Interfaces** | | | | | |
| 10Base-X | No (See note 3.) | 5.3.2.4.2 | 1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to AS-SIP trunk. |
| 100Base-X | No (See note 3.) | 5.3.2.4.2 | 1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3u. Applies to AS-SIP trunk. |
| 1000Base-X | No (See note 3.) | 5.3.2.4.2 | 1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3z and 802.3ab. Applies to AS-SIP trunk. |
| ISDN T1 PRI ANSI T1.619a | Yes | 5.3.2.4.3 | 2, 3, 7, 8, 10, 13 | Certified | Met threshold CRs/FRs . Provides legacy DSN and TELEPORT connectivity. |
| ISDN T1 PRI NI-2 | Yes | 5.3.2.4.3 | 2, 3, 7, 8, 10, 13 | Certified | Met threshold CRs/FRs. Provides PSTN Connectivity |
| T1 CCS7 ANSI T1.619a | No | 5.3.2.12.9 | 2 , 3, 7, 8, 10, 13 | Not Tested | This interface is not offered by the SUT. |
| T1 CAS | No | 5.3.2.12.11 | 2, 3, 7, 8, 10, 13 | Not Tested | This interface is not offered by the SUT. |
| E1 PRI ITU-T Q.955.3 | No (See note 4.) | 5.3.2.12.10 | 2, 3, 7, 8, 10, 13 | Not Tested | This interface is not offered by the SUT. |
| E1 PRI ITU-T Q.931 | No (See note 4.) | 5.3.2.12.10 | 2, 3, 7, 8, 10, 13 | Not Tested | This interface is not offered by the SUT. |
| **NM** | | | | | |
| 10Base-X | No (See note 3.) | 5.3.2.4.4 5.3.2.7.2.8 | 16, 17 | Certified | Met threshold CRs/FRs. Verified via LoC. |
| 100Base-X | No (See note 3.) | 5.3.2.4.4 5.3.2.7.2.8 | 16, 17 | Certified | Met threshold CRs/FRs. Verified via LoC. |

## Table 1. SUT Interface Interoperability Status (continued)

**NOTES:**
1. CR/FR requirements are contained in Table 2. CR/FR numbers represent a roll-up of UCR requirements. Enclosure 3 provides a list of more detailed requirements LSC products.
2. Paragraph 11 of Enclosure 2 provides detailed information pertaining to open TDRs and associated operational impacts.
3. Must provide a minimum of one of the listed interfaces.
4. This interface is conditionally required for deployment in Europe.

**LEGEND:**

| | |
|---|---|
| ANSI | American National Standards Institute |
| ASD NII | Assistant Secretary of Defense for Networks and Information Integration |
| BRI | Basic Rate Interface |
| CAS | Channel Associated Signaling |
| CCS7 | Common Channel Signaling 7 |
| CR | Capability Requirement |
| E1 | 2048 Mbps European trunk standard |
| FR | Functional Requirement |
| IAD | Integrated Access Device |
| ID | Identification |
| IEEE | Institute of Electrical and Electronics Engineers |
| ISDN | Integrated Services Digital Network |
| ITU-T | International Telecommunications Union – Telecommunication Standardization Sector |
| LoC | Letter of Compliance |
| NI-2 | National ISDN-2 |
| NM | Network Management |
| PEI | Proprietary End Instrument |
| PRI | Primary Rate Interface |
| SUT | System Under Test |
| T1 | 1.544 Mbps North American trunk standard |
| TDR | Test Discrepancy Report |
| UCR | Unified capabilities Requirements |

## Table 2. SUT Capability Requirements and Functional Requirements Status

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Status | Remarks |
|---|---|---|---|---|---|
| 1 | **Assured Services Product Features and Capabilities** | | | | |
| | DSCP Packet Marking | Required | 5.3.2.2.1.4 | Met | |
| | Voice Features and Capabilities | Required | 5.3.2.2.2.1 | Partially Met | See note 2. |
| | Public Safety Features | Required | 5.3.2.2.2.2 | Met | |
| | ASAC – Open Loop | Required | 5.3.2.2.2.3 | Met | |
| | Signaling Protocols | Required | 5.3.2.2.3 | Met | |
| | Signaling Performance | Conditional | 5.3.2.2.4 | Met | |
| 2 | **Registration, Authentication, and Failover** | | | | |
| | Registration | Required | 5.3.2.3.1 | Met | |
| | Failover | Required | 5.3.2.3.2 | Met | |
| 3 | **Product Physical, Quality, and Environmental Factors** | | | | |
| | Availability | Required | 5.3.2.5.2.1 | Met | |
| | Maximum Downtimes | Required | 5.3.2.5.2.2 | Met | |
| | Loss of Packets | Required (See note 3.) | 5.3.2.5.4 | Met | |
| 4 | **Voice End Instruments** | | | | |
| | Tones and Announcements | Required | 5.3.2.6.1.1 | Partially Met | See notes 2 and 4. |
| | Audio Codecs | Required | 5.3.2.6.1.2 | Partially Met | See note 4. |
| | VoIP PEI or AEI Audio Performance | Required | 5.3.2.6.1.3 | Partially Met | See note 4. |
| | VoIP Sampling Standard | Required | 5.3.2.6.1.4 | Partially Met | See note 4. |
| | Authentication to LSC | Required | 5.3.2.6.1.5 | Partially Met | See note 4. |
| | Analog Telephone Support | Required (See note 5.) | 5.3.2.6.1.6 | Partially Met | See notes 4 and 6. |
| | Softphones | Conditional | 5.3.2.6.1.7 | Partially Met | See notes 4 and 7. |
| | ISDN BRI | Conditional | 5.3.2.6.1.8 | Not Tested | |

**Table 2. SUT Capability Requirements and Functional Requirements Status (continued)**

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Status | Remarks |
|---|---|---|---|---|---|
| 5 | **Video End Instruments** | | | | |
| | Video End Instrument | Required | 5.3.2.6.2 | Partially Met | See note 8. |
| | Display Messages, Tones, and Announcements | Required | 5.3.2.6.2.1 | Partially Met | See note 8. |
| | Video Codecs (Including Associated Audio Codecs) | Required | 5.3.2.6.2.2 | Partially Met | See note 8. |
| 6 | **LSC Requirements** | | | | |
| | PBAS/ASAC Requirements | Required | 5.3.2.7.2.1 | Met | |
| | Calling Number Delivery Requirements | Required | 5.3.2.7.2.2 | Met | |
| | LSC Signaling Requirements | Required | 5.3.2.7.2.3 | Met | |
| | Service Requirements under Total Loss of WAN Transport | Required | 5.3.2.7.2.4 | Met | |
| | Local Location Server and Directory | Required | 5.3.2.7.2.5 | Met | |
| | LSC Transport Interface Functions | Required | 5.3.2.7.2.7 | Met | |
| | LSC to PEI, AEI, and Operator Console Status Verification | Required | 5.3.2.7.2.10 | Partially Met | See note 9. |
| | Line-Side Custom Features Interference | Conditional | 5.3.2.7.2.11 | Met | |
| | Loop Avoidance | Required (See note 3.) | 5.3.2.7.3 | Met | |
| 7 | **Call Connection Agent Requirements** | | | | |
| | CCA IWF Component | Required (See note 10.) | 5.3.2.9.2.1 | Met | See note 11. |
| | CCA MGC Component | Required | 5.3.2.9.2.2 | Met | |
| | SG Component | Conditional | 5.3.2.9.2.3 | Not Tested | |
| | CCA-IWF Support for AS-SIP | Required | 5.3.2.9.5.1 | Met | |
| | CCA-IWF Support for SS7 | Conditional | 5.3.2.9.5.2 | Not Tested | |
| | CCA-IWF Support for PRI via MG | Required | 5.3.2.9.5.3 | Met | |
| | CCA-IWF Support for CAS Trunks via MG | Conditional | 5.3.2.9.5.4 | Not Tested | |
| | CCA-IWF Support for PEI and AEI Signaling Protocols | Required | 5.3.2.9.5.5 | Partially Met | See note 12. |
| | CCA-IWF Support for VoIP and TDM Protocol Interworking | Required (See note 10.) | 5.3.2.9.5.6 | Met | See note 11. |
| | CCA Preservation of Call Ringing State during Failure Conditions | Required (See note 3.) | 5.3.2.9.6 | Met | |
| | CCA Interactions with Transport Interface Functions | Required | 5.3.2.10.3 | Met | |
| | CCA Interactions with the EBC | Required | 5.3.2.10.4 | Met | |
| | CCA Support for Admission Control | Required | 5.3.2.10.5 | Met | |
| | CCA Support for UFS | Required | 5.3.2.10.6 | Met | |
| | CCA Support for IA | Required | 5.3.2.10.7 | Met | |
| | CCA Interaction with EIs | Required | 5.3.2.10.10 | Partially Met | See notes 7 and 8. |
| | CCA Support for AS Voice and Video | Required | 5.3.2.10.11 | Partially Met | See notes 8 and 9. |
| | CCA Interactions with Service control Functions | Required | 5.3.2.10.12 | Met | |
| | CCA Interworking between AS-SIP and SS7 | Conditional | 5.3.2.11 | Not Tested | |

**Table 2. SUT Capability Requirements and Functional Requirements Status (continued)**

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Status | Remarks |
|---|---|---|---|---|---|
| 8 | **MG Requirements** | | | | |
| | Role of MG In LSC | Required | 5.3.2.12.3.1 | Met | |
| | MG Support for ASAC | Required | 5.3.2.12.4.1 | Met | |
| | MG and IA Functions | Required | 5.3.2.12.4.2 | Met | |
| | MG Interaction with Service Control Function | Required | 5.3.2.12.4.3 | Met | |
| | MG Interactions with IP Transport Interface Functions | Required | 5.3.2.12.4.4 | Met | |
| | MG-EBC interactions | Required | 5.3.2.12.4.5 | Met | |
| | MG IP-Based PSTN Interface Requirements | Conditional | 5.3.2.12.4.7 | Not Tested | |
| | MG Interaction with EIs | Required | 5.3.2.12.4.8 | Partially Met | See note 4. |
| | MG support for User Features and Services | Required | 5.3.2.12.4.9 | Met | |
| | MG Interface to TDM | Required | 5.3.2.12.5 | Met | See note 10. |
| | MG Interface to TDM Allied and Coalition | Conditional | 5.3.2.12.6 | Not Tested | |
| | MG Interface to TDM PSTN in US | Required | 5.3.2.12.7 | Met | See note 11. |
| | MG Interfaces to TDM PSTN OCONUS | Required | 5.3.2.12.8 | Partially Met | See note 12. |
| | MG Support for CCS7 | Conditional | 5.3.2.12.9 | Not Tested | |
| | MG Support for ISDN PRI Trunks | Required | 5.3.2.12.10 | Met | |
| | MG Support for CAS Trunks | Conditional | 5.3.2.12.11 | Not Tested | |
| | MG requirements for VoIP Internal Interfaces | Required | 5.3.2.12.12 | Met | |
| | MG Echo Cancellation | Required | 5.3.2.12.13 | Met | |
| | MG Clock Timing | Required | 5.3.2.12.14 | Met | |
| | MGC-MG CCA Functions | Required | 5.3.2.12.15 | Met | |
| | MG V.150.1 | Required | 5.3.2.12.16 | Not Tested | See note 6. |
| | MG Preservation of Call Ringing during Failure | Required (See note 3.) | 5.3.2.12.17 | Met | |
| 9 | **SG Requirements** | | | | |
| | SG and CCS7 network Interactions | Conditional | 5.3.2.13.5.1 | Not Tested | |
| | SG Interactions with CCA | Conditional | 5.3.2.13.5.2 | Not Tested | |
| | SG Interworking Functions | Conditional | 5.3.2.13.5.3 | Not Tested | |
| 10 | **WWNDP Requirements** | | | | |
| | WWNDP | Required | 5.3.2.16 | Met | |
| | DSN WWNDP | Required | 5.3.2.16.1 | Met | |

**Table 2. SUT Capability Requirements and Functional Requirements Status (continued)**

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Status | Remarks |
|---|---|---|---|---|---|
| 11 | **Commercial Cost Avoidance** | | | | |
| | Commercial Cost Avoidance | Required | 5.3.2.23 | Partially Met | See note 13 |
| 12 | **AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)** | | | | |
| | AS-SIP Requirements for External Interfaces | Conditional | 5.3.2.24 | Not Tested | |
| 13 | **Precedence Call Diversion** | | | | |
| | Precedence Call Diversion | Required | 5.3.2.25 | Met | |
| 14 | **Attendant Station Features** | | | | |
| | Precedence and Preemption | Required (See note 3.) | 5.3.2.26.1 | Not Tested | See note 14. |
| | Call Display | Required (See note 3.) | 5.3.2.26.2 | Not Tested | See note 14. |
| | Class of Service Override | Required (See note 3.) | 5.3.2.26.3 | Not Tested | See note 14.. |
| | Busy Override and Busy Verification | Required (See note 3.) | 5.3.2.26.4 | Not Tested | See note 14. |
| | Night service | Required (See note 3.) | 5.3.2.26.5 | Not Tested | See note 14. |
| | Automatic Recall of Attendant | Required (See note 3.) | 5.3.2.26.6 | Not Tested | See note 14. |
| | Calls in Queue to the Attendant | Required (See note 3.) | 5.3.2.26.7 | Not Tested | See note 14. |
| 15 | **AS-SIP Requirements** | | | | |
| | SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs | Required (See note 3.) | 5.3.4.7 | Not Tested | See note 4. |
| | SIP Session Keep-Alive Timer | Required | 5.3.4.8 | Met | |
| | Session Description Protocol | Required | 5.3.4.9 | Met | |
| | Precedence and Preemption | Required | 5.3.4.10 | Met | |
| | Video Telephony – General Rules | Required | 5.3.4.12 | Not Met | See note 8. |
| | Calling Services | Required | 5.3.4.13 | Met | |
| | SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances | Required | 5.3.4.14 | Met | |
| | Relevant Timers for the Terminating Gateway and the Originating Gateway | Required | 5.3.4.15 | Met | |
| | SIP Requirements for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.16 | Met | |
| | Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.17 | Met | |
| | Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.18 | Met | |
| | Supplementary Services | Required | 5.3.4.19 | Met | |
| 16 | **IPv6 Requirements** | | | | |
| | Product Requirements | Required | 5.3.5.4 | Partially Met | See note 13. |
| 17 | **NM** | | | | |
| | LSC Management Function | Required | 5.3.2.7.2.6 | Partially Met | See note 15. |
| | VVoIP NMS Interface Requirements | Required | 5.3.2.4.4 | Partially Met | See note 15. |
| | General Management requirements | Required | 5.3.2.17.2 | Partially Met | See note 15. |
| | Requirement for FCAPS Management | Required | 5.3.2.17.3 | Partially Met | See notes 15 and 16. |
| | NM requirements of Appliance Functions | Required | 5.3.2.18 | Partially Met | See note 15. |
| | Accounting Management | Required | 5.3.2.19 | Partially Met | See note 15. |

**Table 2. SUT Capability Requirements and Functional Requirements Status (continued)**

**NOTES:**
1. Annotation of 'required' refers to high level requirement category. Applicability of each sub-requirement is provided in enclosure 3.
2. The SUT had outstanding open TDRs at the completion of testing adjudicated by DISA to have a minor operational impact. The vendor has submitted a PoAM to address the open TDRs. Paragraph 11 of Enclosure 2 provides additional details.
3. This requirement represents a new UCR requirement where the vendor has 18-months (July 2011) to comply.
4. SUT met the requirement for PEIs; SUT was not tested with generic AEI requirements because no AEI was provided. AEIs are a new UCR 2008 Change 1 requirement; the vendor has 18-months (July 2011) to comply.
5. UCR 2008 Change 1 added 18-month rule for V.150.1 IAD support.
6. Vendor did not demonstrate the V.150.1 support but has until July 2011 to comply with this requirement.
7. Met both voice and video requirements via Softphone.
8. Demonstrated video requirements via Softphone only, not PEIs (Proprietary Hard Video Phones). Vendor did not provide a PEI video capability. This was adjudicated by DISA to have a low operational impact because of the limited deployment of PEIs with video.
9. SUT partially met PEI requirements (no video). The AEI and Operator Console requirements were not tested; the 18-month rule for complying (July 2011) applies.
10. The SUT must meet T1 PRI (T1.619a and NI-2) IWF. T1 CAS and T1 CCS7 interfaces are conditional.
11. The SUT met T1 PRI ((T1.619a and NI-2) IWF requirements. The T1 CAS and T1 CCS7 interfaces were not supported by the SUT.
12. The SUT met PEI CCA-IWF requirements. The AEI CCA-IWF requirements were not tested. The 18-month rule applies to AEIs.
13. The Vendor submitted an IPv6 LoC with noted discrepancies which include the interface for Commercial Cost Avoidance functionality. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.
14. The Attendant Console requirements are new UCR requirements; 18-month rule applies.
15. The SUT submitted an NM LoC with noted discrepancies. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.
16. SUT does not support destination code controls. This was adjudicated by DISA to have a low operational impact because of the limited deployment of users.

**LEGEND:**

| | | | |
|---|---|---|---|
| AEI | AS-SIP End Instrument | LSC | Local Session Controller |
| AS | Assured Services | MG | Media Gateway |
| ASAC | Assured Services Admission Control | MGC | Media Gateway Controller |
| AS-SIP | Assured Services Session Initiation Protocol | NM | Network Management |
| BRI | Basic Rate Interface | NMS | Network Management System |
| CAS | Channel Associated Signaling | OCONUS | Outside the Continental United States |
| CCA | Call Connection Agent | PBAS | Precedence-Based Assured Service |
| CCS7 | Common Channel Signaling 7 | PEI | Proprietary End Instrument |
| CR | Capabilities Requirement | PoAM | Plan of Actions and Milestones |
| DSCP | Differentiated Services Code Point | PRI | Primary Rate Interface |
| DSN | Defense Switched Network | PSTN | Public Switch Telephone Network |
| EBC | Edge Boundary Controller | SG | Signaling Gateway |
| EI | End Instrument | SIP | Session Initiation Protocol |
| FCAPS | Fault, Configuration, Accounting, Performance, and Security | SS7 | Signaling System Number 7 |
| | | SUT | System Under Test |
| FR | Functional Requirement | T1 | 1.544 Mbps North American trunk standard |
| IA | Information Assurance | TDM | Time Division Multiplexing |
| IAD | Integrated Access Device | TDR | Test Discrepancy Report |
| ID | Identification | UCR | Unified Capabilities Requirements |
| IP | Internet Protocol | UFS | User Features and Services |
| IPv6 | Internet Protocol version 6 | VoIP | Voice over Internet Protocol |
| ISDN | Integrated Services Digital Network | VVoIP | Voice and Video over Internet Protocol |
| IWF | Interworking Function | WAN | Wide Area Network |
| LoC | Letter of Compliance | WWNDP | World Wide Numbering and Dialing Plan |

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at https://jit.fhu.disa.mil (NIPRNet).

Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. All associated data is available on the Defense Information Systems Agency Unified Capability Coordination Office (UCCO) website located at http://www.disa.mil/ucco/.

6. The JITC point of contact is Stephane Arsenault, JITC, commercial (520) 538-5269 or DSN 312-879-5269; e-mail address is Stephane.Arsenault@ disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number is 0911801.

FOR THE COMMANDER:

Steve Laszeski

3 Enclosures a/s

for RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

Distribution (electronic mail):
Joint Staff J-6
Joint Interoperability Test Command, Liaison, TE3/JT1
Office of Chief of Naval Operations, CNO N6F2
Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)
Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
U.S. Marine Corps MARCORSYSCOM, SIAT, MJI Division I
DOT&E, Net-Centric Systems and Naval Warfare
U.S. Coast Guard, CG-64
Defense Intelligence Agency
National Security Agency, DT
Defense Information Systems Agency, TEMC
Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

## ADDITIONAL REFERENCES

(c) Office of the Assistant Secretary of Defense, “Department of Defense Unified Capabilities Requirements 2008, Change 1,” 22 January 2010

(d) Joint Interoperability Test Command, “Unified Capabilities Test Plan (UCTP),” Draft

(e) Joint Interoperability Test Command, “Information Assurance (IA) Assessment of Avaya Aura™ AS5300 Local Session Controller, Version 2.0 (TN 0911801),” 31 March 2011

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## CERTIFICATION TESTING SUMMARY

**1. SYSTEM TITLE.** The Avaya Aura™ AS5300 Local Session Controller (LSC), Version 2.0 (with specified patch releases), hereinafter referred to as the System Under Test (SUT).

**2. SPONSOR. .** Defense Information Systems Agency, Attention: Louis Schmuckler GS15, Capability Center, IP Division Chief, GS23, PO Box 4502, Arlington VA, 22204-4502, Phone 703-882-0274, e-mail: louis.schmuckler@disa.mil.

**3. SYSTEM POC**. Avaya Government Solutions, Attention: Scott Birdzell, Address: 12730 Fair Lakes Circle, Fairfax, VA, 22033-4901, Phone: 480-282-1066, e-mail: Scott.Birdzell@avayagov.com.

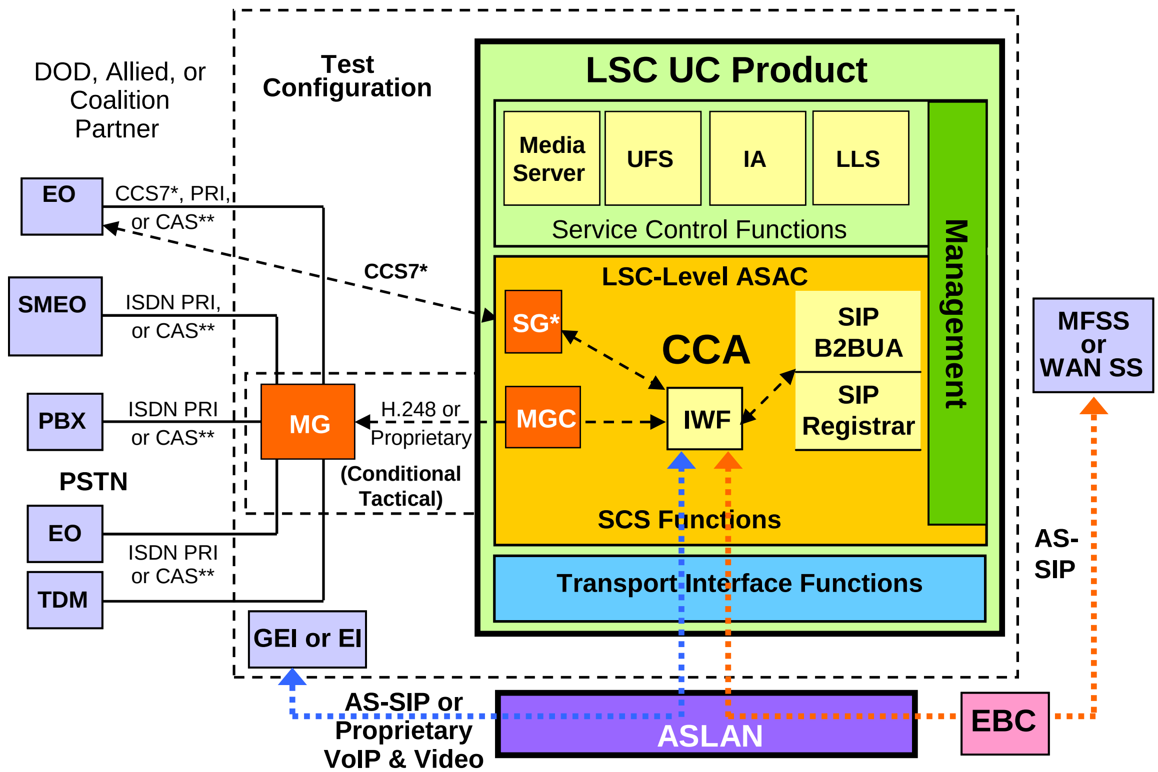
**4. TESTER.** Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

**5. SYSTEM DESCRIPTION.** The SUT is a software-based call processing product that provides voice and video services which includes legacy 2-wire telephones, Internet Protocol (IP) telephones, and media processing devices within a local service domain. Additionally, an LSC extends signaling and call control services to allow calls to reach connections outside the local service domain. The LSC software and functions may be distributed physically among several high-availability server platforms with redundant call management modules and subscriber tables to provide robustness.

The SUT is a secure Unified Communications (UC) system using Transport Layer Security *(*TLS*)* and Secure Real-time Transport Protocol (SRTP) originally designed and manufactured by Nortel (now acquired by Avaya). The system provides integrated IP telephony, conferencing, and voice mail that meets, Department of Defense (DoD) security and service assurance requirements. The AS5300 is used for creating, modifying, and terminating two-party (unicast) or multiparty (multicast) media streams, supporting up to 25,000 subscribers per system.

The SUT is a session manager designed to increase productivity and collaboration by allowing users to collaborate in an integrated solution. The SUT supports the Session Initiation Protocol (SIP), Assured Services – SIP (AS-SIP), Secure Real-time Transport Protocol, Session Description Protocol, Security Descriptions for Media Streams, Transport Layer Security, Nortel encrypted Unified Networks Internet Protocol Stimulus (UNIStim), Multilevel Precedence and Preemption (MLPP) using American National Standards Institute (ANSI) Primary Rate Interface (PRI) with T1.619a. Services that are included are: instant messaging (IM), presence, voice, video calling, internal voicemail, audio ad hoc conferencing, audio meet me conferencing, Messaging Application Server (MAS)-Music on Hold (MOH), and MAS-Unified Communications (UCOMM).

**6. OPERATIONAL ARCHITECTURE.** Figure 2-1 depicts the LSC functional model and Figure 2-2 the notional operational architecture that the SUT may be used in.

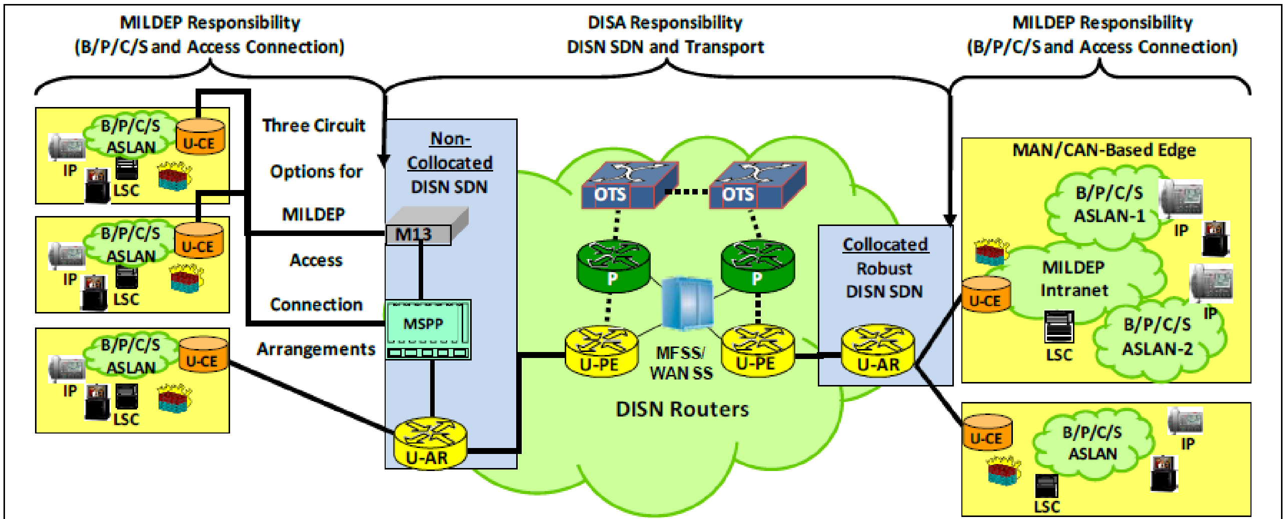



*Support for an SG & CCS7 is Conditional for an LSC.
**Support for CAS is Conditional for an LSC.
NOTE: The EI (using Proprietary VoIP & Video) is part of the LSC UC product, the GEI (AS-SIP VoIP & Video) is not.

**LEGEND:**

| | | | |
|---|---|---|---|
| ASAC | Assured Service Admission Control | LSC | Local Session Controller |
| ASLAN | Assured Services Local Area Network | MFS | Multifunction Switch |
| AS-SIP | Assured Services Session Initiation Protocol | MFSS | Multifunction Softswitch |
| B2BUA | Back-to-Back User Agent | MG | Media Gateway |
| CAS | Channel Associated Signaling | MGC | Media Gateway Controller |
| CCA | Call Connection Agent | PBX | Private Branch Exchange |
| CCS7 | Common Channel Signaling No. 7 | PRI | Primary Rate Interface |
| DOD | Department of Defense | PSTN | Public Switched Telephone Network |
| EBC | Edge Border Controller | SCS | Session Control and Signaling |
| EI | End Instrument | SG | Signaling Gateway |
| EO | End Office | SIP | Session Initiation Protocol |
| GEI | Generic End Instrument | SS | Softswitch |
| IA | Information Assurance | TDM | Time Division Multiplexer |
| ISDN | Integrated Services Digital Network | UC | Unified Capabilities |
| IWF | Interworking Function | UFS | User Features and Services |
| LLS | Local Location Service | WAN | Wide Area Network |
| LSC | Local Session Controller | VoIP | Voice over Internet Protocol |

**Figure 2-1. LSC Functional Reference Model**




**LEGEND:**

| | | | |
|---|---|---|---|
| ASLAN | Assured Services Local Area Network | MILDEP | Military Department |
| B/P/C/S | Base, Post, Camp, Station | MSPP | Multi-Service Provisioning Platform |
| CAN | Campus Area Network | OTS | Optical Transport System |
| DISA | Defense Information Systems Agency | P | Provider |
| DISN | Defense Information System Network | SDN | Service Delivery Node |
| IP | Internet Protocol | SS | Softswitch |
| LSC | Local Session Controller | U-AR | Unclassified Aggregation Router |
| M13 | Multiplexer | U-CE | Unclassified Customer Edge Router |
| MAN | Metropolitan Area Network | U-PE | Unclassified Provider Edge Router |
| MFSS | Multifunction Softswitch | WAN | Wide Area Network |

**Figure 2-2. UC Network Architecture**

**7. INTEROPERABILITY REQUIREMENTS.** The interface, Capability Requirements (CR) and Functional Requirements (FR), Information Assurance (IA), and other requirements for LSCs are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of Reference (c).

**7.1 Interfaces.** The SUT uses the external interfaces to connect to the Global Information Grid (GIG) network and other Unified Capabilities products. Table 2-1, shows the physical interfaces supported by the SUT. The table documents the physical interfaces and the associated standards.

**Table 2-1. LSC Interface Requirements**

| Interface | Critical | UCR Reference | Criteria | Remarks |
|---|---|---|---|---|
| **Line Interfaces** | | | | |
| 10Base-X | Yes | 5.3.2.6.3 | Support minimum threshold CRs/FRs (2, 4,10, 13, and 16) and meet interface criteria for 802.3i and 802.3j | |
| 100Base-X | Yes | 5.3.2.6.3 | Support minimum threshold CRs/FRs (2, 4,10, 13, and 16) and meet interface criteria for 802.3.u | |

## Table 2-1. LSC Interface Requirements (continued)

| Interface | Critical | UCR Reference | Criteria (See note 1.) | Remarks |
|---|---|---|---|---|
| **Line Interfaces (continued)** | | | | |
| 1000Base-X | No | 5.3.2.6.3 | Support minimum threshold CRs/FRs (2, 4, 10, 13, and 16) and meet interface criteria for 802.3z. | |
| 2-wire analog | Yes | 5.3.2.6.1.6 | Support minimum threshold CRs/FRs (2, 4, 10, and 13) and meet interface criteria for analog. | |
| BRI | No | 5.3.2.6.1.8 | Support minimum threshold CRs/FRs (2, 4, 10, and 13) and meet interface criteria for BRI | |
| **External Interfaces** | | | | |
| 10Base-X | No (See note 2.) | 5.3.2.4.2 | Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16) and meet interface criteria for 802.3i and 802.3j | |
| 100Base-X | No (See note 2.) | 5.3.2.4.2 | Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16) and meet interface criteria for 802.3u | |
| 1000Base-X | No (See note 2.) | 5.3.2.4.2 | Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16)and meet interface criteria for 802.3z | |
| ISDN T1 PRI ANSI T1.619a | Yes | 5.3.2.4.3 | Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for ISDN T1 PRI (T1.619a) | Provides legacy DSN and TELEPORT connectivity. |
| ISDN T1 PRI NI-2 | Yes | 5.3.2.4.3 | Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13)and meet interface criteria for ISDN T1 PRI (NI-2) | Provides PSTN Connectivity. |
| T1 CCS7 ANSI T1.619a | No | 5.3.2.12.9 | Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for T1 CCS7 (ANSI T1.619a) | |
| T1 CAS | No | 5.3.2.12.11 | Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for T1 CAS | T1 CAS with MLPP. |
| E1 PRI ITU-T Q.955.3 | No | 5.3.2.12.10 | Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13)and meet interface criteria for EI PRI (Q.955.3) | Conditionally required for DSN European connectivity. |
| E1 PRI ITU-T Q.931 | No | 5.3.2.12.10 | Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13)and meet interface criteria for E1 PRI (ITU-T Q.931) | Conditionally required for commercial European connectivity. |
| **NM** | | | | |
| 10Base-X | No (See note 2.) | 5.3.2.4.4<br>5.3.2.7.2.8 | Support minimum threshold CR/FRs (16 and 17) and meet interface criteria for 802.3i and 802.3j | |
| 100Base-X | No (See note 2.) | 5.3.2.4.4<br>5.3.2.7.2.8 | Support minimum threshold CR/FRs (16 and 17) and meet interface criteria for 802.3u | |

**NOTES:**
1. CR/FR requirements are contained in Table 2-2. CR/FR numbers represent a roll-up of UCR requirements. Enclosure 3 provides a list of more detailed requirements for security device products.
2. Must provide a minimum of one of the listed interfaces.

**LEGEND:**

| | | | |
|---|---|---|---|
| ANSI | American National Standards Institute | ITU-T | International Telecommunication Union – Telecommunication Standardization Sector |
| BRI | Basic Rate Interface | LSC | Local Session Controller |
| CR | Capability Requirement | Mbps | Megabits per second |
| CCS7 | Common Channel Signaling | MLPP | Multi-Level Precedence and Preemption |
| DSN | Defense Switched Network | NI-2 | National ISDN Standard 2 |
| E1 | European Basic Multiplex Rate (2.048 Mbps) | PRI | Primary Rate Interface |
| FR | Functional Requirement | PSTN | Public Switched Telephone Network |
| ISDN | Integrated Services Digital Network | T1 | Digital Transmission Link Level 1 (1.544 Mbps) |
| | | UCR | Unified Capabilities Requirements |

**7.2 Capability Requirements (CR) and Functional Requirements (FR).** The LSCs have required and conditional features and capabilities that are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of the UCR. The SUT does not need to provide non-critical (conditional) requirements. If they are provided, they must function according to the specified requirements. The SUTs features and capabilities and its aggregated requirements are listed in Table 2-2. Detailed CR/FR requirements are provided in Table 3-1 of Enclosure 3.

**Table 2-2. LSC Capability Requirements and Functional Requirements**

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Criteria |
|---|---|---|---|---|
| 1 | **Assured Services Product Features and Capabilities** | | | |
| | DSCP Packet Marking | Required | 5.3.2.2.1.4 | See note 2. |
| | Voice Features and Capabilities | Required | 5.3.2.2.2.1 | |
| | Public Safety Features | Required | 5.3.2.2.2.2 | |
| | ASAC – Open Loop | Required | 5.3.2.2.3 | |
| | Signaling Protocols | Required | 5.3.2.2.2.3 | |
| | Signaling Performance | Required | 5.3.2.2.4 | |
| 2 | **Registration, Authentication, and Failover** | | | |
| | Registration | Required | 5.3.2.3.1 | See note 2. |
| | Failover | Required | 5.3.2.3.2 | |
| 3 | **Product Physical, Quality, and Environmental Factors** | | | |
| | Availability | Required | 5.3.2.5.2.1 | See note 2. |
| | Maximum Downtimes | Required | 5.3.2.5.2.2 | |
| | Loss of Packets | Required (See note 3.) | 5.3.2.5.4 | |
| 4 | **Voice End Instruments** | | | |
| | Tones and Announcements | Required | 5.3.2.6.1.1 | See note 2. |
| | Audio Codecs | Required | 5.3.2.6.1.2 | |
| | VoIP PEI or AEI Audio Performance Requirements | Required | 5.3.2.6.1.3 | |
| | VoIP Sampling Standard | Required | 5.3.2.6.1.4 | |
| | Authentication To LSC | Required | 5.3.2.6.1.5 | |
| | Analog Telephone Support | Required (See note 4.) | 5.3.2.6.1.6 | |
| | Softphones | Conditional | 5.3.2.6.1.7 | |
| | ISDN BRI | Conditional | 5.3.2.6.1.8 | |
| 5 | **Video End Instruments** | | | |
| | Video End Instrument | Required | 5.3.2.6.2 | See note 2. |
| | Display Messages, Tones, and Announcements | Required | 5.3.2.6.2.1 | |
| | Video Codecs (Including Associated Audio Codecs) | Required | 5.3.2.6.2.2 | |
| 6 | **LSC Requirements** | | | |
| | PBAS/ASAC Requirements | Required | 5.3.2.7.2.1 | See note 2. |
| | Calling Number Delivery Requirements | Required | 5.3.2.7.2.2 | |
| | LSC Signaling Requirements | Required | 5.3.2.7.2.3 | |
| | Service Requirements under Total Loss of WAN Transport | Required | 5.3.2.7.2.4 | |
| | Local Location Server and Directory | Required | 5.3.2.7.2.5 | |
| | LSC Transport Interface Functions | Required | 5.3.2.7.2.7 | |
| | LSC to PEI, AEI, and Operator Console Status Verification | Required | 5.3.2.7.2.10 | |
| | Line-Side Custom Features Interference | Conditional | 5.3.2.7.2.11 | |
| | Loop Avoidance | Required (See note 3.) | 5.3.2.7.3 | |

## Table 2-2. LSC Capability Requirements and Functional Requirements (continued)

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Criteria |
|---|---|---|---|---|
| 7 | **Call Connection Agent Requirements** | | | |
| | CCA IWF Component | Required (See note 5.) | 5.3.2.9.2.1 | See note 2. |
| | CCA MGC Component | Required (See note 5.) | 5.3.2.9.2.2 | |
| | SG Component | Conditional | 5.3.2.9.2.3 | |
| | CCA-IWF Support for AS-SIP | Required | 5.3.2.9.5.1 | |
| | CCA-IWF Support for SS7 | Conditional | 5.3.2.9.5.2 | |
| | CCA-IWF Support for PRI via MG | Required | 5.3.2.9.5.3 | |
| | CCA-IWF Support for CAS Trunks via MG | Conditional | 5.3.2.9.5.4 | |
| | CCA-IWF Support for PEI and AEI Signaling Protocols | Required | 5.3.2.9.5.5 | |
| | CCA-IWF Support for VoIP and TDM Protocol Interworking | Required (See note 5.) | 5.3.2.9.5.6 | |
| | CCA Preservation of Call Ringing State during Failure Conditions | Required | 5.3.2.9.6 | |
| | CCA Interactions with Transport Interface Functions | Required | 5.3.2.10.3 | |
| | CCA Interactions with the EBC | Required | 5.3.2.10.4 | |
| | CCA Support for Admission Control | Required | 5.3.2.10.5 | |
| | CCA Support for UFS | Required | 5.3.2.10.6 | |
| | CCA Support for IA | Required | 5.3.2.10.7 | |
| | CCA Interaction with EIs | Required | 5.3.2.10.10 | |
| | CCA Support for AS Voice and Video | Required | 5.3.2.10.11 | |
| | CCA Interactions with Service control Functions | Required | 5.3.2.10.12 | |
| | CCA Interworking between AS-SIP and SS7 | Conditional | 5.3.2.11 | |
| 8 | **MG Requirements** | | | |
| | Role of MG In LSC | Required | 5.3.2.12.3.1 | See note 2. |
| | MG Support for ASAC | Required | 5.3.2.12.4.1 | |
| | MG and IA Functions | Required | 5.3.2.12.4.2 | |
| | MG Interaction with Service Control Function | Required | 5.3.2.12.4.3 | |
| | MG Interactions with IP Transport Interface Functions | Required | 5.3.2.12.4.4 | |
| | MG-EBC interactions | Required | 5.3.2.12.4.5 | |
| | MG IP-Based PSTN Interface Requirements | Conditional | 5.3.2.12.4.7 | |
| | MG Interaction with EIs | Required | 5.3.2.12.4.8 | |
| | MG support for User Features and Services | Required | 5.3.2.12.4.9 | |
| | MG Interface to TDM | Required (See note 6.) | 5.3.2.12.5 | |
| | MG Interface to TDM Allied and Coalition | Conditional | 5.3.2.12.6 | |
| | MG Interface to TDM PSTN in US | Required (See note 6.) | 5.3.2.12.7 | |
| | MG Interfaces to TDM PSTN OCONUS | Required (See notes 6 and 7.) | 5.3.2.12.8 | |
| | MG Support for CCS7 | Conditional | 5.3.2.12.9 | |

## Table 2-2. LSC Capability Requirements and Functional Requirements (continued)

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Criteria |
|---|---|---|---|---|
| 8 | **MG Requirements (continued)** | | | |
| | MG Support for ISDN PRI Trunks | Required | 5.3.2.12.10 | See note 2. |
| | MG Support for CAS Trunks | Required | 5.3.2.12.11 | |
| | MG requirements for VoIP Internal Interfaces | Required | 5.3.2.12.12 | |
| | MG Echo Cancellation | Required | 5.3.2.12.13 | |
| | MG Clock Timing | Required | 5.3.2.12.14 | |
| | MGC-MG CCA Functions | Required | 5.3.2.12.15 | |
| | MG V.150.1 | Required | 5.3.2.12.16 | |
| | MG Preservation of Call Ringing during Failure | Required (See note 3.) | 5.3.2.12.17 | |
| 9 | **SG Requirements** | | | |
| | SG and CCS7 network Interactions | Conditional | 5.3.2.13.5.1 | See note 2. |
| | SG Interactions with CCA | Conditional | 5.3.2.13.5.2 | |
| | SG Interworking Functions | Conditional | 5.3.2.13.5.3 | |
| 10 | **WWNDP Requirements** | | | |
| | WWNDP | Required | 5.3.2.16 | See note 2. |
| | DSN WWNDP | Required | 5.3.2.16.1 | |
| 11 | **Commercial Cost Avoidance** | | | |
| | Commercial Cost Avoidance | Required (See note 3.) | 5.3.2.23 | See note 2. |
| 12 | **AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)** | | | |
| | AS-SIP Requirements for External Interfaces | Conditional | 5.3.2.24 | See note 2. |
| 13 | **Precedence Call Diversion** | | | |
| | Precedence call Diversion | Required | 5.3.2.25 | See note 2. |
| 14 | **Attendant Station Features** | | | |
| | Precedence and Preemption | Required (See note 3.) | 5.3.2.26.1 | See note 2. |
| | Call Display | Required (See note 3.) | 5.3.2.26.2 | |
| | Class of Service Override | Required (See note 3.) | 5.3.2.26.3 | |
| | Busy Override and Busy Verification | Required (See note 3.) | 5.3.2.26.4 | |
| | Night service | Required (See note 3.) | 5.3.2.26.5 | |
| | Automatic Recall of Attendant | Required (See note 3.) | 5.3.2.26.6 | |
| | Calls in Queue to the Attendant | Required (See note 3.) | 5.3.2.26.7 | |
| 15 | **AS-SIP Requirements** | | | |
| | SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs | Required (See note 3.) | 5.3.4.7 | See note 2. |
| | SIP Session Keep-Alive Timer | Required | 5.3.4.8 | |
| | Session Description Protocol | Required | 5.3.4.9 | |
| | Precedence and Preemption | Required | 5.3.4.10 | |
| | Video Telephony – General Rules | Required | 5.3.4.12 | |
| | Calling Services | Required | 5.3.4.13 | |

## Table 2-2. LSC Capability Requirements and Functional Requirements (continued)

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Criteria |
|---|---|---|---|---|
| 15 | **AS-SIP Requirements (continued)** | | | |
| | SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances | Required | 5.3.4.14 | See note 2. |
| | Relevant Timers for the Terminating Gateway and the Originating Gateway | Required | 5.3.4.15 | |
| | SIP Requirements for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.16 | |
| | Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.17 | |
| | Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.18 | |
| | Supplementary Services | Required | 5.3.4.19 | |
| 16 | **IPv6 Requirements** | | | |
| | Product Requirements | Required | 5.3.5.4 | See note 2. |
| 17 | **NM** | | | |
| | LSC Management Function | Required | 5.3.2.7.2.6 | See note 2. |
| | VVoIP NMS Interface Requirements | Required | 5.3.2.4.4 | |
| | General Management requirements | Required | 5.3.2.17.2 | |
| | Requirement for FCAPS Management | Required | 5.3.2.17.3 | |
| | NM requirements of Appliance Functions | Required | 5.3.2.18 | |
| | Accounting Management | Required | 5.3.2.19 | |

**NOTES:**

1. Annotation of 'required' refers to high level requirement category. Applicability of each sub-requirement is provided in Enclosure 3.
2. Detailed requirements and associated criteria for LSC s are listed in Table 3-1 of Enclosure 3.
3. This requirement represents a new UCR requirement for which the vendor has 18-months (July 2011) to comply.
4. The UCR 2008 Change 1 added 18-month rule for G.711 and V.150.1 IAD support.
5. The LSC must meet T1 PRI (ANSI T1.619a and NI-2) CCA IWF. The T1 CAS and T1 CCS7 CCA IWF are conditional.
6. The LSC must meet TDM requirements for T1 PRI (ANSI T1.619a and NI-2). The TDM requirements for T1 CAS and T1 CCS7 are conditional.
7. The E1 requirements for OCOUNUS are conditionally required for deployments in Europe.

## Table 2-2. LSC Capability Requirements and Functional Requirements (continued)

**LEGEND:**

| | | | |
|---|---|---|---|
| AEI | AS-SIP End Instrument | LoC | Letter of Compliance |
| AS | Assured Services | LSC | Local Session Controller |
| ASAC | Assured Services Admission Control | MG | Media Gateway |
| AS-SIP | Assured Services Session Initiation Protocol | MGC | Media Gateway Controller |
| BRI | Basic Rate Interface | NM | Network Management |
| CAS | Channel Associated Signaling | NMS | Network Management System |
| CCA | Call Connection Agent | OCONUS | Outside the Continental United States |
| CCS7 | Common Channel Signaling 7 | PBAS | Precedence-Based Assured Service |
| CR | Capabilities Requirement | PEI | Proprietary End Instrument |
| DSCP | Differentiated Services Code Point | PSTN | Public Switch Telephone Network |
| DSN | Defense Switched Network | SG | Signaling Gateway |
| EBC | Edge Boundary Controller | SIP | Session Initiation Protocol |
| EI | End Instrument | SS7 | Signaling System Number 7 |
| FCAPS | Fault, Configuration, Accounting, Performance, and Security | SUT | System Under Test |
| | | T1 | 1.544 Mbps North American trunk standard |
| FR | Functional Requirement | TDM | Time Division Multiplexing |
| IA | Information Assurance | UCR | Unified Capabilities Requirements |
| IAD | Integrated Access Device | UFS | User Features and Services |
| ID | Identification | VoIP | Voice over Internet Protocol |
| IP | Internet Protocol | VVoIP | Voice and Video over Internet Protocol |
| IPv6 | Internet Protocol version 6 | WAN | Wide Area Network |
| ISDN | Integrated Services Digital Network | WWNDP | World Wide Numbering and Dialing Plan |
| IWF | Interworking Function | | |

**7.3 Information Assurance.** Table 2-3 details the Information Assurance (IA) requirements applicable to an LSC.

## Table 2-3. LSC IA Requirements

| Requirement | Applicability (See note.) | UCR Reference | Criteria |
|---|---|---|---|
| General Requirements | Required | 5.4.6.2 | Detailed requirements and associated criteria for LSC are listed in the IATP (Reference (e)). |
| Authentication | Required | 5.4.6.2.1 | |
| Integrity | Required | 5.4.6.2.2 | |
| Confidentiality | Required | 5.4.6.2.3 | |
| Non-Repudiation | Required | 5.4.6.2.4 | |
| Availability | Required | 5.4.6.2.5 | |

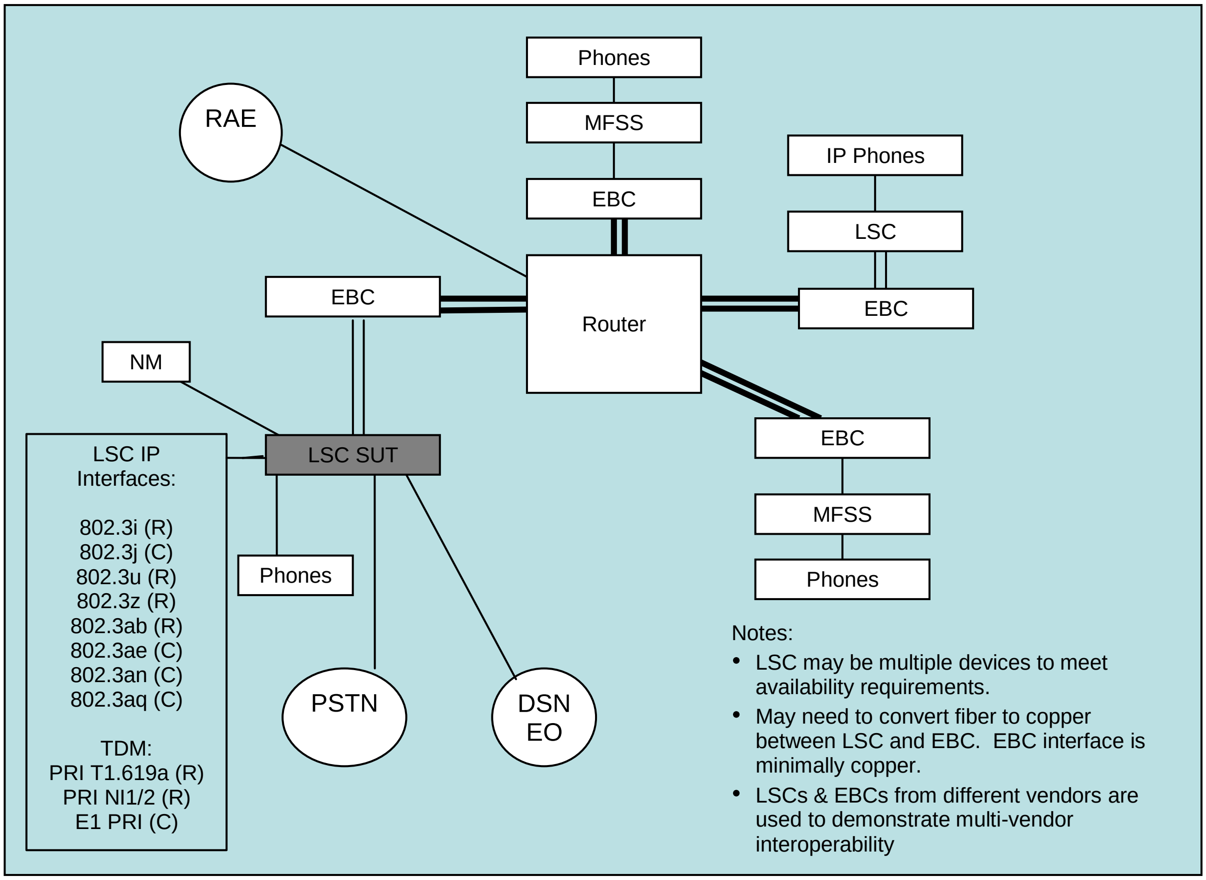
**NOTE:** Annotation of ‘required’ refers to high level requirement category. Applicability of each sub-requirement is provided in Reference (e).

**LEGEND:**

| | | | |
|---|---|---|---|
| IA | Information Assurance | LSC | Local Session Controller |
| IATP | IA Test Plan | UCR | Unified capabilities Requirements |

**7.4 Other.** None

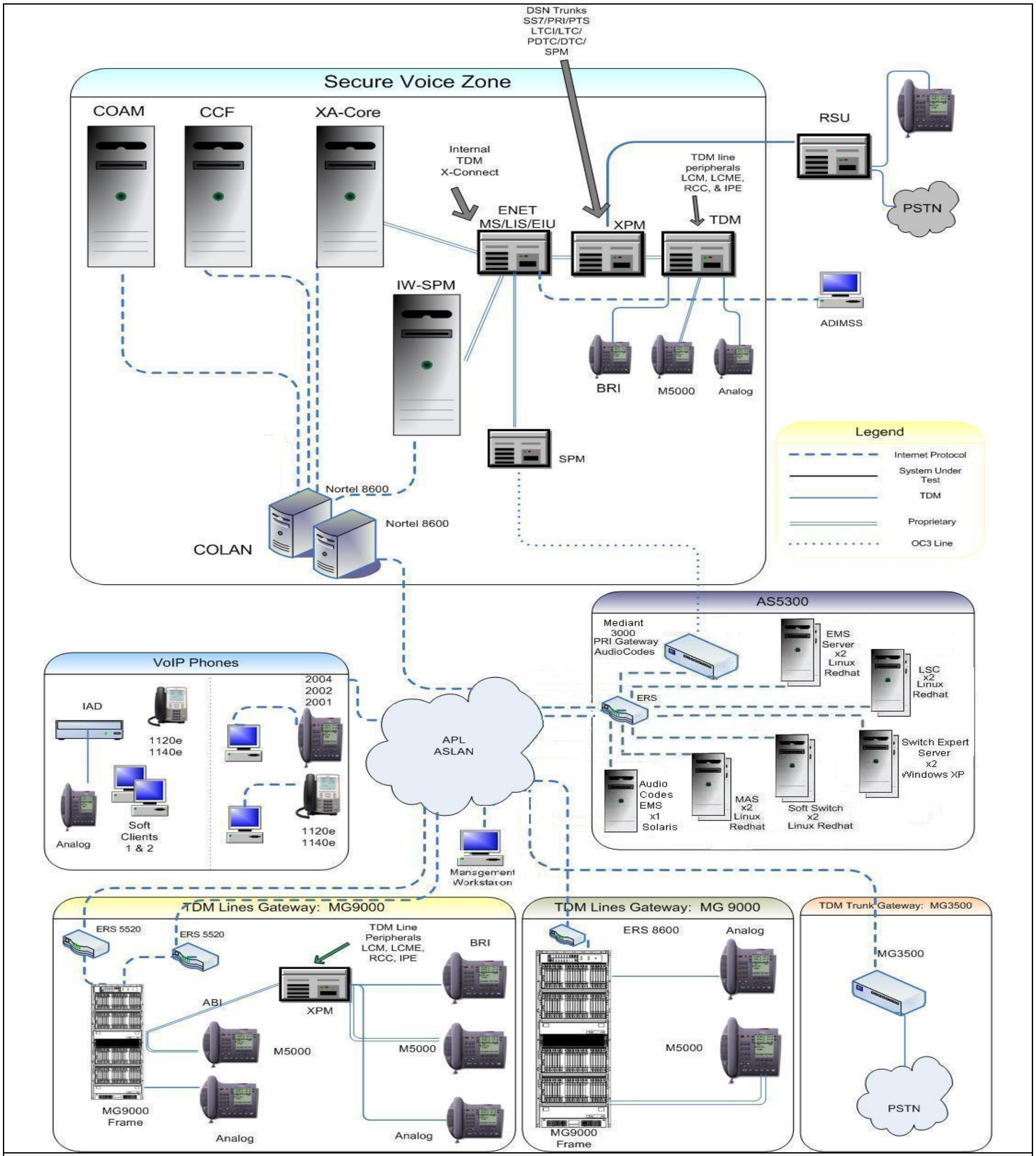
**8. TEST NETWORK DESCRIPTION.** The SUT was tested at JITC in a manner and configuration similar to that of a notional operational environment. Testing the system's required functions and features was conducted using the test configurations depicted in Figures 2-3 and 2-4. Figure 2-3 depicts the minimum test architecture for testing LSCs. Figure 2-4 depicts the SUT's test configuration.




**LEGEND:**

| | | | |
|---|---|---|---|
| C | Conditional | NM | Network Management |
| DSN | Defense Switched Network | PSTN | Public Switch Telephone Network |
| EBC | Edge Boundary Controller | R | Required |
| EO | End Office | RAE | Required Ancillary Equipment |
| IP | Internet Protocol | SUT | System Under Test |
| LSC | Local Session Controller | TDM | Time Division Multiplexing |
| MFSS | MultiFunction Softswitch | | |

**Figure 2-3. LSC Minimum Test Architecture**




**LEGEND:**

| | | | |
|---|---|---|---|
| APL | Approved Products List | M3K | M3000 |
| ASLAN | Assured Services Local Area Network | Mgr | Manager |
| DSN | Defense Switched Network | PRI | Primary Rate Interface |
| EMS | Element Management System | PSTN | Public Switch Telephone Network |
| ERS | Ethernet Router Switch | RTS | Real Time Services |
| IAD | Integrated Access Device | SIP | Session Initiation Protocol |
| IP | Internet Protocol | TDM | Time Division Multiplexing |
| LSC | Local Session Controller | | |

**Figure 2-4. AS5300 LSC Test Configuration**

**LEGEND:**

| | |
|---|---|
| ADIMSS | Advanced DSN Integrated Management Support System |
| APL | Approved Products List |
| ASLAN | Assured Services Local Area Network |
| BRI | Basic Rate Interface |
| CCF | Call Control Frame |
| COAM | Centralized Operation, Administration and Maintenance |
| COLAN | Central Office LAN |
| CS2100 | Communications Server 21st Century |
| DSN | Defense Switched Network |
| EIU | Ethernet Interface Unit |
| EMS | Element Management System |
| ENET | Enhanced Network |
| ERS | Ethernet Router Switch |
| IAD | Integrated Access Device |
| IP | Internet Protocol |
| IPE | Intelligent Peripheral Module |
| LAN | Local Area Network |
| LCM | Line Concentration Module |
| LCME | Line Concentration Module Enhanced |
| LIS | Link Interface Shelf |
| LSC | Local Session Controller |
| M3K | M3000 |
| MAS | Media Application Server |
| MG | Media Gateway |
| Mgr | Manager |
| MS | Message Switch |
| OC3 | Optical Carrier Level 3 |
| PRI | Primary Rate Interface |
| PSTN | Public Switch Telephone Network |
| RCC | Remote Cluster Controller |
| RTS | Real Time Services |
| SIP | Session Initiation Protocol |
| SPM | Spectrum Peripheral Module |
| TDM | Time Division Multiplexing |
| XA-Core | Extended Architecture Core |
| XPM | XMS-Based Peripheral Module |
| XMS | Extended Multiprocessor System |

**Figure 2-4. AS5300 LSC Test Configuration (continued)**

**9. SYSTEM CONFIGURATIONS.** Table 2-4 provides the system configurations and hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine its interoperability capability with associated network devices and network traffic.

## Table 2-4. Tested System Configurations

| System Name | Software Release |
|---|---|
| Avaya CS2100 | Succession Enterprise (SE) 09.1 |
| Siemens EWSD | 19d with Patch Set 46 |
| **SUT Local Session Controller** | |
| Avaya AS5300 | AS5300 Release 2.0<br>Load: MCP_13.0.0.0_2010-05-05-2108<br>Patch: MCP_13.0.0.6_2010-11-30-1031.patch |

| **SUT Components** | | |
|---|---|---|
| **SIP Core OAM&P** | | |
| **Part Number** | **Part Description** | **Firmware/Software** |
| NTVW02AD | AS 5300 Release 2.0 SIP Core New System Software Package | Load: MCP_13.0.0.5_2010-10-07-1208, patch: MCP_13.0.0.6_2010-11-30-1031.patch<br>Oracle version: 10.2.0.4.0, patch level 19 |
| NTVW02AF | AS 5300 R1.0 to R2.0 Upgrade w/SRS PrePaid | Load: MCP_13.0.0.0_2010-05-05-2108<br>Patch: MCP_13.0.0.5_2010-10-07-1208 patch<br>Oracle version: 10.2.0.4.0, patch level 19 |
| NTVW02AG | AS 5300 R1.0 to R2.0 Upgrade and Expansions | Load: MCP_13.0.0.5_2010-10-07-1208, patch: MCP_13.0.0.6_2010-11-30-1031.patch<br>Oracle version: 10.2.0.4.0, patch level 19 |
| NTVW02DE | AS 5300 Release 2.0 OS Software Kit – CDROM and DVD | Version: 13.0.13<br>Patch to: 13.0.18 |
| NTVW02DD | AS 5300 Release 2.0 Core Apps Software Kit CDROM and DVD | Load: MCP_13.0.0.5_2010-10-07-1208, patch: MCP_13.0.0.6_2010-11-30-1031.patch<br>Oracle version: 10.2.0.4.0, patch level 19 |
| **SIP Core LSC Session Manager** | | |
| **Part Number** | **Part Description** | **Firmware/Software** |
| NTVW02AD | AS 5300 Release 2.0 SIP Core New System Software Package | Load: MCP_13.0.0.5_2010-10-07-1208, patch: MCP_13.0.0.6_2010-11-30-1031.patch |
| NTVW02AF | AS 5300 R1.0 to R2.0 Upgrade w/SRS PrePaid | Load: MCP_13.0.0.5_2010-10-07-1208, patch: MCP_13.0.0.6_2010-11-30-1031.patch |
| NTVW02AG | AS 5300 R1.0 to R2.0 Upgrade and Expansions | Load: MCP_13.0.0.5_2010-10-07-1208, patch: MCP_13.0.0.6_2010-11-30-1031.patch |
| NTVW02DE | AS 5300 Release 2.0 OS Software Kit – CDROM and DVD | Version: 13.0.21 |
| NTVW02DD | AS 5300 Release 2.0 Core Apps Software Kit CDROM and DVD | Load: MCP_13.0.0.5_2010-10-07-1208, patch: MCP_13.0.0.6_2010-11-30-1031.patch |
| **Media Application Server** | | |
| **Part Number** | **Part Description** | **Firmware/Software** |
| NTVW02AE | AS 5300 Release 2.0 MAS New System Software Package | Version: 6.6.0.74 |
| NTVW02AF | AS 5300 R1.0 to R2.0 Upgrade w/SRS PrePaid | Version: 6.6.0.74 |
| NTVW02AG | AS 5300 R1.0 to R2.0 Upgrade and Expansions | Version: 6.6.0.74 |
| NTVW02DF | AS 5300 Rls 2.0 MAS Base and Applications | Version: 6.6.0.74 |
| NTVW02DE | AS 5300 Release 2.0 OS Software Kit – CDROM and DVD | Version: 13.0.19 |
| **Audiocodes PRI Gateway** | | |
| **Part Number** | **Part Description** | **Firmware/Software** |
| NTVW02CF | AS 5300 Release 2.0 Audiocodes M3K PRI GW 8 Span Bundle DC | Version: 5.80A.045.000 |
| NTVW02CG | AS 5300 Release 2.0 Audiocodes M3K PRI GW 8 Span Bundle AC | Version: 5.80A.045.000 |
| NTVW02CH | AS 5300 Release 2.0 Audiocodes M3K PRI GW 12 Span Bundle DC | Version: 5.80A.045.000 |
| NTVW02CI | AS 5300 Release 2.0 Audiocodes M3K PRI GW 12 Span Bundle AC | Version: 5.80A.045.000 |
| NTVW02CJ | AS 5300 Release 2.0 Audiocodes M3K PRI GW 16 Span Bundle DC | Version: 5.80A.045.000 |

## Table 2-4. Tested System Configurations (continued)

| Local Session Controller (continued) | | |
|---|---|---|
| **Audiocodes PRI Gateway (continued)** | | |
| **Part Number** | **Part Number** | **Part Number** |
| NTVW02CK | AS 5300 Release 2.0 Audiocodes M3K PRI GW 16 Span Bundle AC | Version: 5.80A.045.000 |
| NTVW02CL | AS 5300 Release 2.0 Audiocodes M3K PRI GW 42 Span Bundle DC | Version: 5.80A.045.000 |
| NTVW02CM | AS 5300 Release 2.0 Audiocodes M3K PRI GW 42 Span Bundle AC | Version: 5.80A.045.000 |
| NTVW02DG | AS 5300 Release 2.0 Audiocodes M3K PRI GW Software CD | Version: 5.80A.045.000 |
| NTVW02CN | AS 5300 Release 2.0 Audiocodes M3K PRI GW OC3 Bundle DC | Version: 5.80A.045.000 |
| NTVW02CO | AS 5300 Release 2.0 Audiocodes M3K PRI GW OC3 Bundle AC | Version: 5.80A.045.000 |
| NTVW02DH | AS 5300 Release 2.0 Audiocodes M3K/6310 PRI GW Software CD | Version: 5.80A.045.000 |
| **Audiocodes Element Management System** | | |
| **Part Number** | **Part Description** | **Firmware/Software** |
| NTVW02AY | AS5300 Audiocodes EMS Server Package AC | Version: 5.8.83 |
| NTVW02AZ | AS5300 Audiocodes EMS Server Package DC | |
| NTVW02DK | AS5300 Release 2.0 Audiocodes EMS Application SW on Sun Netra T2000 Server with Oracle 9.2 and Solaris 10 and JAVA ES installed | |
| **Device Name** | | **Software Release** |
| CPE AS 5300 UC Client (previously MMPC Client) | | Version: 7.2.3068_20101007 |
| AS 5300 1120E and 1140E ASSIP IP Deskphones | | SIP Based Version: 03.01.09.00 |
| Audiocodes MP112 (2 port Integrated Access Device) | | Version: 5.80A.054.001 |
| Audiocodes MP124 (24 port Integrated Access Device) | | Version: 5.80A.054.001 |
| Management Workstation | | Customer Provided STIG'd PC |

**LEGEND:**

| | |
|---|---|
| AC | Alternating Current |
| Apps | Applications |
| AS | Assured Services |
| CDROM | Compact Disk Read-Only Memory |
| CPE | Customer Premise Equipment |
| DC | Direct Current |
| DVD | Digital Video Disk |
| EMS | Element Management System |
| EWSD | Elektronisches Wählsystem Digital |
| GW | Gateway |
| IP | Internet Protocol |
| LSC | Local Session Controller |
| M3K | M3000 |
| MAS | Media Application Server |
| MMPC | Multimedia Personal Computer |
| OC3 | Optical Carrier-Level 3 |
| OS | Operating System |
| PC | Personal Computer |
| PRI | Primary Rate Interface |
| SIP | Session Initiation Protocol |
| STIG | Security Technical Implementation Guide |
| SUT | System Under Test |
| SW | Software |
| UC | Unified Capabilities |
| UNISTIM | Unified Networks IP Stimulus |

**10. TESTING LIMITATIONS.** The JITC test team noted the following testing limitations including the impact they may have on interpretation of the results and conclusions. Any untested requirements are also included in the testing limitations.

**a. Call Loading.** The JITC could not create a large volume of line calls because the line signaling protocol used by the SUT is proprietary. Also, the JITC could not generate a huge volume of AS-SIP trunk calls because of the limited lines available and lack of AS-SIP test equipment. These limitations pose a low risk to interoperability and should not impact overall results and conclusions. The use of operational data as the LSC is fielded will validate the SUT's ability to support its proposed number of subscribers (up to 25,000).

**b. Assured Services Session Initiation Protocol End Instruments.** The JITC did not test the SUT with generic AS-SIP End Instruments (AEIs) because none were available at the time of test. A vendor has yet to submit a product as an AEI for certification. This requirement was an addition to the UCR 2008 Change 1 and therefore the SUT has 18-months (July 2011) to comply with the requirement.

**c. Proprietary End Instruments.** The JITC did not test Proprietary End Instruments (PEIs) for video requirements. The only devices tested for video were Softphones. The video phones were tested intra-enclave and to the Avaya Multifunction Softswitch. Multi-vendor interoperability has not been demonstrated. Since the Defense Switched Network (DSN) has not deployed videophones under legacy certifications, this poses a low operational impact.

**d. Internet Protocol version 6.** The JITC tested the SUT to verify its Internet Protocol version 6 (IPv6) capabilities intra-enclave. The vendor submitted a letter of compliance (LoC) stating limitations. Paragraph 11 provides detailed information about Ipv6 results.

**e. Network Management.** The JITC did not test the SUT's Network Management (NM) capabilities to meet UCR requirements. The vendor did submit an NM LoC that was reviewed by JITC. The JITC's evaluation of the SUT's NM capabilities is provided in paragraph 11.

**f. Attendant Consoles.** The JITC did not test the SUT's Attendant features. The vendor did not provide an Attendant Console. This requirement was an addition to the UCR 2008 Change 1 and therefore the SUT has 18-months (July 2011) to comply with the requirement.

**g. Master/Slave.** The JITC did not test the SUT to determine its ability to meet master/slave requirements. Initial fielding of an LSC will not be used in this configuration. The DISA adjudicated the discrepancy as having a low operational impact.

**h. Secure Data and Secure Voice Calls.** Since the standard for modem over IP is based on International Telecommunication Union -Telecommunication Standardization Sector (ITU-T) V.150.1 and vendors have 18-months (July 2011) to comply, secure calls could not be tested inter-enclave (between LSCs via the Defense Information System Network (DISN)).

**11. INTEROPERABILITY EVALUATION RESULTS.** The SUT meets the critical interoperability requirements for an LSC in accordance with the UCR and is certified for joint use with other UC Products listed on the UC Approved Products List (APL). Additional discussion regarding specific testing results is located in subsequent paragraphs.

**11.1 Interfaces.** The SUT met line interface requirements for 10/100 Base-X interfaces. These IP line interfaces were met through use of PEIs (voice only) and Softphone clients (voice and video). The SUT supports 2-wire analog phones via an Integrated Access Device (IAD). The SUT met the external interface requirements for 10/100/100Base-X (AS-SIP) and Integrated Services Digital Network (ISDN) PRI for both ANSI T1.619a MLPP and National ISDN-2 (NI-2) commercial. The JITC did not test the other conditional interfaces. The interface status of the SUT is provided in Table 2-5.

**Table 2-5. SUT Interface Requirements Status**

| Interface | Critical | UCR Reference | Threshold CR/FR Requirements (See note 1.) | Status | Remarks (See note 2.) |
|---|---|---|---|---|---|
| **Line Interfaces** | | | | | |
| 10Base-X | Yes | 5.3.2.6.3 | 2, 4, 10, 13, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to PEIs (voice) and Softphones (voice and video). |
| 100Base-X | Yes | 5.3.2.6.3 | 2, 4, 10, 13, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3u. Applies to PEIs (voice) and Softphones (voice and video). |
| 1000Base-X | No | 5.3.2.6.3 | 2, 4, 10,13, 16 | Not Tested | This interface is not offered by the SUT PEIs. |
| 2-wire analog | Yes | 5.3.2.6.1.6 | 2, 4, 10, 13, | Certified | Met threshold CRs/FRs for 2-wire instruments. Applies to 2-wire secure and non-secure analog instruments. Requirement met through use of an IAD that supports IEEE 802.3i, 802.3u, and 802.3ab. |
| BRI | No | 5.3.2.6.1.8 | 2, 4, 10, 13 | Not Tested | This interface is not supported by the SUT. |
| **External Interfaces** | | | | | |
| 10Base-X | No (See note 3.) | 5.3.2.4.2 | 1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to AS-SIP trunk. |
| 100Base-X | No (See note 3.) | 5.3.2.4.2 | 1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3u. Applies to AS-SIP trunk. |
| 1000Base-X | No (See note 3.) | 5.3.2.4.2 | 1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16 | Certified | Met threshold CRs/FRs for IEEE 802.3z and 802.3ab. Applies to AS-SIP trunk. |
| ISDN T1 PRI ANSI T1.619a | Yes | 5.3.2.4.3 | 2, 3, 7, 8, 10, 13 | Certified | Met threshold CRs/FRs . Provides legacy DSN and TELEPORT connectivity. |
| ISDN T1 PRI NI-2 | Yes | 5.3.2.4.3 | 2, 3, 7, 8, 10, 13 | Certified | Met threshold CRs/FRs. Provides PSTN Connectivity |
| T1 CCS7 ANSI T1.619a | No | 5.3.2.12.9 | 2 , 3, 7, 8, 10, 13 | Not Tested | This interface is not offered by the SUT. |
| T1 CAS | No | 5.3.2.12.11 | 2, 3, 7, 8, 10, 13 | Not Tested | This interface is not offered by the SUT. |

## Table 2-5. SUT Interface Requirements Status (continued)

| Interface | Critical | UCR Reference | Threshold CR/FR Requirements (See note 1.) | Status | Remarks (See note 2.) |
|---|---|---|---|---|---|
| E1 PRI ITU-T Q.955.3 | No | 5.3.2.12.10 | 2, 3, 7, 8, 10, 13 | Not Tested | This interface is not offered by the SUT. |
| E1 PRI ITU-T Q.931 | No | 5.3.2.12.10 | 2, 3, 7, 8, 10, 13 | Not Tested | This interface is not offered by the SUT. |
| **NM** | | | | | |
| 10Base-X | No (Note 3) | 5.3.2.4.4<br>5.3.2.7.2.8 | 16, 17 | Certified | Met threshold CRs/FRs. Verified via LoC. |
| 100Base-X | No (Note 3) | 5.3.2.4.4<br>5.3.2.7.2.8 | 16, 17 | Certified | Met threshold CRs/FRs. Verified via LoC. |

**NOTES:**
1. CR/FR requirements are contained in Table 2. CR/FR numbers represent a roll-up of UCR requirements. Enclosure 3 provides a list of more detailed requirements LSC products.
2. Paragraph 11 of Enclosure 2 provides detailed information pertaining to open TDRs and associated operational impacts.
3. Must provide a minimum of one of the listed interfaces.

**LEGEND:**

| | |
|---|---|
| ANSI | American National Standards Institute |
| ASD NII | Assistant Secretary of Defense for Networks and Information Integration |
| BRI | Basic Rate Interface |
| CAS | Channel Associated Signaling |
| CCS7 | Common Channel Signaling 7 |
| CR | Capability Requirement |
| E1 | 2048 Mbps European trunk standard |
| FR | Functional Requirement |
| IAD | Integrated Access Device |
| ID | Identification |
| IEEE | Institute of Electrical and Electronics Engineers |
| ISDN | Integrated Services Digital Network |
| ITU-T | International Telecommunications Union – Telecommunication Standardization Sector |
| LoC | Letter of Compliance |
| NI-2 | National ISDN-2 |
| NM | Network Management |
| PEI | Proprietary End Instrument |
| PRI | Primary Rate Interface |
| SUT | System Under Test |
| T1 | 1.544 Mbps North American trunk standard |
| TDR | Test Discrepancy Report |
| UCR | Unified capabilities Requirements |

**11.2 Capability Requirements (CR) and Functional Requirements (FR).** The SUT CR and FR status is depicted in Table 2-6. Detailed CR/FR requirements are provided in Enclosure 3, Table 3-1. A summary of the SUT's ability to meet UCR requirements are provided in the sub-paragraphs below. All requirements and associated references were derived from UCR 2008 Change 1. Discrepancies discussed below were adjudicated to be minor based on vendor submission and compliance to a Plan of Actions and Milestones.

## Table 2-6. SUT Capability Requirements and Functional Requirements Status

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Status | Remarks |
|---|---|---|---|---|---|
| 1 | **Assured Services Product Features and Capabilities** | | | | |
| | DSCP Packet Marking | Required | 5.3.2.2.1.4 | Met | |
| | Voice Features and Capabilities | Required | 5.3.2.2.2.1 | Partially Met | See note 2. |
| | Public Safety Features | Required | 5.3.2.2.2.2 | Met | |
| | ASAC – Open Loop | Required | 5.3.2.2.2.3 | Met | |
| | Signaling Protocols | Required | 5.3.2.2.3 | Met | |
| | Signaling Performance | Conditional | 5.3.2.2.4 | Met | |
| 2 | **Registration, Authentication, and Failover** | | | | |
| | Registration | Required | 5.3.2.3.1 | Met | |
| | Failover | Required | 5.3.2.3.2 | Met | |

## Table 2-6. SUT Capability Requirements and Functional Requirements Status (continued)

| CR/FR ID | Capability/ Function | Applicability (See note 1.) | UCR Reference | Status | Remarks |
|---|---|---|---|---|---|
| 3 | **Product Physical, Quality, and Environmental Factors** | | | | |
| | Availability | Required | 5.3.2.5.2.1 | Met | |
| | Maximum Downtimes | Required | 5.3.2.5.2.2 | Met | |
| | Loss of Packets | Required (See note 3.) | 5.3.2.5.4 | Met | |
| 4 | **Voice End Instruments** | | | | |
| | Tones and Announcements | Required | 5.3.2.6.1.1 | Partially Met | See notes 2 and 4. |
| | Audio Codecs | Required | 5.3.2.6.1.2 | Partially Met | See note 4. |
| | VoIP PEI or AEI Audio Performance | Required | 5.3.2.6.1.3 | Partially Met | See note 4. |
| | VoIP Sampling Standard | Required | 5.3.2.6.1.4 | Partially Met | See note 4. |
| | Authentication to LSC | Required | 5.3.2.6.1.5 | Partially Met | See note 4. |
| | Analog Telephone Support | Required (See note 5.) | 5.3.2.6.1.6 | Partially Met | See notes 4 and 6. |
| | Softphones | Conditional | 5.3.2.6.1.7 | Partially Met | See notes 4 and 7. |
| | ISDN BRI | Conditional | 5.3.2.6.1.8 | Not Tested | |
| 5 | **Video End Instruments** | | | | |
| | Video End Instrument | Required | 5.3.2.6.2 | Partially Met | See note 8. |
| | Display Messages, Tones, and Announcements | Required | 5.3.2.6.2.1 | Partially Met | See note 8. |
| | Video Codecs (Including Associated Audio Codecs) | Required | 5.3.2.6.2.2 | Partially Met | See note 8. |
| 6 | **LSC Requirements** | | | | |
| | PBAS/ASAC Requirements | Required | 5.3.2.7.2.1 | Met | |
| | Calling Number Delivery Requirements | Required | 5.3.2.7.2.2 | Met | |
| | LSC Signaling Requirements | Required | 5.3.2.7.2.3 | Met | |
| | Service Requirements under Total Loss of WAN Transport | Required | 5.3.2.7.2.4 | Met | |
| | Local Location Server and Directory | Required | 5.3.2.7.2.5 | Met | |
| | LSC Transport Interface Functions | Required | 5.3.2.7.2.7 | Met | |
| | LSC to PEI, AEI, and Operator Console Status Verification | Required | 5.3.2.7.2.10 | Partially Met | See note 9. |
| | Line-Side Custom Features Interference | Conditional | 5.3.2.7.2.11 | Met | |
| | Loop Avoidance | Required (See note 3.) | 5.3.2.7.3 | Met | |
| 7 | **Call Connection Agent Requirements** | | | | |
| | CCA IWF Component | Required (See note 10.) | 5.3.2.9.2.1 | Met | See note 11. |
| | CCA MGC Component | Required | 5.3.2.9.2.2 | Met | |
| | SG Component | Conditional | 5.3.2.9.2.3 | Not Tested | |
| | CCA-IWF Support for AS-SIP | Required | 5.3.2.9.5.1 | Met | |
| | CCA-IWF Support for SS7 | Conditional | 5.3.2.9.5.2 | Not Tested | |
| | CCA-IWF Support for PRI via MG | Required | 5.3.2.9.5.3 | Met | |
| | CCA-IWF Support for CAS Trunks via MG | Conditional | 5.3.2.9.5.4 | Not Tested | |
| | CCA-IWF Support for PEI and AEI Signaling Protocols | Required | 5.3.2.9.5.5 | Partially Met | See note 12. |
| | CCA-IWF Support for VoIP and TDM Protocol Interworking | Required (See note 10.) | 5.3.2.9.5.6 | Met | See note 11. |
| | CCA Preservation of Call Ringing State during Failure Conditions | Required (See note 3.) | 5.3.2.9.6 | Met | |
| | CCA Interactions with Transport Interface Functions | Required | 5.3.2.10.3 | Met | |
| | CCA Interactions with the EBC | Required | 5.3.2.10.4 | Met | |

## Table 2-6. SUT Capability Requirements and Functional Requirements Status (continued)

| CR/FR ID | Capability/Function | Applicability (See note 1) | UCR Reference | Status | Remarks |
|---|---|---|---|---|---|
| 7 | **Call Connection Agent Requirements (continued)** | | | | |
| | CCA Support for Admission Control | Required | 5.3.2.10.5 | Met | |
| | CCA Support for UFS | Required | 5.3.2.10.6 | Met | |
| | CCA Support for IA | Required | 5.3.2.10.7 | Met | |
| | CCA Interaction with EIs | Required | 5.3.2.10.10 | Partially Met | See notes 7 and 8. |
| | CCA Support for AS Voice and Video | Required | 5.3.2.10.11 | Partially Met | See notes 8 and 9. |
| | CCA Interactions with Service control Functions | Required | 5.3.2.10.12 | Met | |
| | CCA Interworking between AS-SIP and SS7 | Conditional | 5.3.2.11 | Not Tested | |
| 8 | **MG Requirements** | | | | |
| | Role of MG In LSC | Required | 5.3.2.12.3.1 | Met | |
| | MG Support for ASAC | Required | 5.3.2.12.4.1 | Met | |
| | MG and IA Functions | Required | 5.3.2.12.4.2 | Met | |
| | MG Interaction with Service Control Function | Required | 5.3.2.12.4.3 | Met | |
| | MG Interactions with IP Transport Interface Functions | Required | 5.3.2.12.4.4 | Met | |
| | MG-EBC interactions | Required | 5.3.2.12.4.5 | Met | |
| | MG IP-Based PSTN Interface Requirements | Conditional | 5.3.2.12.4.7 | Not Tested | |
| | MG Interaction with EIs | Required | 5.3.2.12.4.8 | Partially Met | See note 4. |
| | MG support for User Features and Services | Required | 5.3.2.12.4.9 | Met | |
| | MG Interface to TDM | Required | 5.3.2.12.5 | Met | See note 10. |
| | MG Interface to TDM Allied and Coalition | Conditional | 5.3.2.12.6 | Not Tested | |
| | MG Interface to TDM PSTN in US | Required | 5.3.2.12.7 | Met | See note 11. |
| | MG Interfaces to TDM PSTN OCONUS | Required | 5.3.2.12.8 | Met | See note 12. |
| | MG Support for CCS7 | Conditional | 5.3.2.12.9 | Not Tested | |
| | MG Support for ISDN PRI Trunks | Required | 5.3.2.12.10 | Met | |
| | MG Support for CAS Trunks | Conditional | 5.3.2.12.11 | Not Tested | |
| | MG requirements for VoIP Internal Interfaces | Required | 5.3.2.12.12 | Met | |
| | MG Echo Cancellation | Required | 5.3.2.12.13 | Met | |
| | MG Clock Timing | Required | 5.3.2.12.14 | Met | |
| | MGC-MG CCA Functions | Required | 5.3.2.12.15 | Met | |
| | MG V.150.1 | Required (See note 3.) | 5.3.2.12.16 | Not Tested | See note 6. |
| | MG Preservation of Call Ringing during Failure | Required (See note 3.) | 5.3.2.12.17 | Not Tested | |
| 9 | **SG Requirements** | | | | |
| | SG and CCS7 network Interactions | Conditional | 5.3.2.13.5.1 | Not Tested | |
| | SG Interactions with CCA | Conditional | 5.3.2.13.5.2 | Not Tested | |
| | SG Interworking Functions | Conditional | 5.3.2.13.5.3 | Not Tested | |
| 10 | **WWNDP Requirements** | | | | |
| | WWNDP | Required | 5.3.2.16 | Met | |
| | DSN WWNDP | Required | 5.3.2.16.1 | Met | |
| 11 | **Commercial Cost Avoidance** | | | | |
| | Commercial Cost Avoidance | Required | 5.3.2.23 | Partially Met | See note 13. |

## Table 2-6. Avaya Aura AS5300 Capability Requirements and Functional Requirements (continued)

| CR/FR ID | Capability/Function | Applicability (See note 1.) | UCR Reference | Status | Remarks |
|---|---|---|---|---|---|
| 12 | **AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)** | | | | |
| | AS-SIP Requirements for External Interfaces | Conditional | 5.3.2.24 | Not Tested | |
| 13 | **Precedence Call Diversion** | | | | |
| | Precedence Call Diversion | Required | 5.3.2.25 | Met | |
| 14 | **Attendant Station Features** | | | | |
| | Precedence and Preemption | Required (See note 3.) | 5.3.2.26.1 | Not Tested | See note 14. |
| | Call Display | Required (See note 3.) | 5.3.2.26.2 | Not Tested | See note 14. |
| | Class of Service Override | Required (See note 3.) | 5.3.2.26.3 | Not Tested | See note 14. |
| | Busy Override and Busy Verification | Required (See note 3.) | 5.3.2.26.4 | Not Tested | See note 14. |
| | Night service | Required (See note 3.) | 5.3.2.26.5 | Not Tested | See note 14. |
| | Automatic Recall of Attendant | Required (See note 3.) | 5.3.2.26.6 | Not Tested | See note 14. |
| | Calls in Queue to the Attendant | Required (See note 3.) | 5.3.2.26.7 | Not Tested | See note 14. |
| 15 | **AS-SIP Requirements** | | | | |
| | SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs | Required (See note 3.) | 5.3.4.7 | Not Tested | See note 4. |
| | SIP Session Keep-Alive Timer | Required | 5.3.4.8 | Met | |
| | Session Description Protocol | Required | 5.3.4.9 | Met | |
| | Precedence and Preemption | Required | 5.3.4.10 | Met | |
| | Video Telephony – General Rules | Required | 5.3.4.12 | Not Met | See note 8. |
| | Calling Services | Required | 5.3.4.13 | Met | |
| | SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances | Required | 5.3.4.14 | Met | |
| | Relevant Timers for the Terminating Gateway and the Originating Gateway | Required | 5.3.4.15 | Met | |
| | SIP Requirements for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.16 | Met | |
| | Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.17 | Met | |
| | Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances | Required | 5.3.4.18 | Met | |
| | Supplementary Services | Required | 5.3.4.19 | Met | |
| 16 | **IPv6 Requirements** | | | | |
| | Product Requirements | Required | 5.3.5.4 | Partially Met | See note 13. |
| 17 | **NM** | | | | |
| | LSC Management Function | Required | 5.3.2.7.2.6 | Partially Met | See note 15. |
| | VVoIP NMS Interface Requirements | Required | 5.3.2.4.4 | Partially Met | See note 15. |
| | General Management requirements | Required | 5.3.2.17.2 | Partially Met | See note 15. |
| | Requirement for FCAPS Management | Required | 5.3.2.17.3 | Partially Met | See notes 15 and 16. |
| | NM requirements of Appliance Functions | Required | 5.3.2.18 | Partially Met | See note 15. |
| | Accounting Management | Required | 5.3.2.19 | Partially Met | See note 16. |

## Table 2-6. Avaya Aura AS5300 Capability Requirements and Functional Requirements (continued)

**NOTES:**
1. Annotation of 'required' refers to high level requirement category. Applicability of each sub-requirement is provided in enclosure 3.
2. The SUT had outstanding open TDRs at the completion of testing adjudicated by DISA to have a minor operational impact. The vendor has submitted a PoAM to address the open TDRs. Paragraph 11 of Enclosure 2 provides additional details.
3. This requirement represents a new UCR requirement where the vendor has 18-months (July 2011) to comply.
4. SUT met the requirement for PEIs; SUT was not tested with generic AEI requirements because no AEI was provided. AEIs are a new UCR 2008 Change 1 requirement; the vendor has 18-months (July 2011) to comply.
5. UCR 2008 Change 1 added 18-month rule for V.150.1 IAD support.
6. Vendor did not demonstrate the V.150.1 support but has until July 2011 to comply with this requirement.
7. Met both voice and video requirements via Softphone.
8. Demonstrated video requirements via Softphone only, not PEIs. Vendor did not provide a PEI video capability. This was adjudicated by DISA to have a low operational impact because of the limited deployment of PEIs with video.
9. SUT partially met PEI requirements (no video). The AEI and Operator Console requirements were not tested; the 18-month rule for complying (July 2011) applies.
10. The SUT must meet T1 PRI (T1.619a and NI-2) IWF. The T1 CAS and T1 CCS7 are conditional.
11. The SUT met T1 PRI ((T1.619a and NI-2) IWF requirements. The T1 CAS and T1 CCS7 were not supported by the SUT.
12. The SUT met PEI CCA-IWF requirements. The AEI CCA-IWF requirements were not tested. The 18-month rule applies to AEIs.
13. The SUT submitted an IPv6 LoC with noted discrepancies which include the interface for Commercial Cost Avoidance functionality. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.
14. The Attendant Console requirements are new UCR requirements; 18-month rule applies.
15. The SUT submitted an NM LoC with noted discrepancies. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.
16. SUT does not support destination code controls. This was adjudicated by DISA to have a low operational impact because of the limited deployment of users.

**LEGEND:**

| | | | |
|---|---|---|---|
| AEI | AS-SIP End Instrument | LSC | Local Session Controller |
| AS | Assured Services | MG | Media Gateway |
| ASAC | Assured Services Admission Control | MGC | Media Gateway Controller |
| AS-SIP | Assured Services Session Initiation Protocol | NM | Network Management |
| BRI | Basic Rate Interface | NMS | Network Management System |
| CAS | Channel Associated Signaling | OCONUS | Outside the Continental United States |
| CCA | Call Connection Agent | PBAS | Precedence-Based Assured Service |
| CCS7 | Common Channel Signaling 7 | PEI | Proprietary End Instrument |
| CR | Capabilities Requirement | PoAM | Plan of Actions and Milestones |
| DSCP | Differentiated Services Code Point | PRI | Primary Rate Interface |
| DSN | Defense Switched Network | PSTN | Public Switch Telephone Network |
| EBC | Edge Boundary Controller | SG | Signaling Gateway |
| EI | End Instrument | SIP | Session Initiation Protocol |
| FCAPS | Fault, Configuration, Accounting, Performance, and Security | SS7 | Signaling System Number 7 |
| | | SUT | System Under Test |
| FR | Functional Requirement | T1 | 1.544 Mbps North American trunk standard |
| IA | Information Assurance | TDM | Time Division Multiplexing |
| IAD | Integrated Access Device | TDR | Test Discrepancy Report |
| ID | Identification | UCR | Unified Capabilities Requirements |
| IP | Internet Protocol | UFS | User Features and Services |
| IPv6 | Internet Protocol version 6 | VoIP | Voice over Internet Protocol |
| ISDN | Integrated Services Digital Network | VVoIP | Voice and Video over Internet Protocol |
| IWF | Interworking Function | WAN | Wide Area Network |
| LoC | Letter of Compliance | WWNDP | World Wide Numbering and Dialing Plan |

### a. Assured Services Product Features and Capabilities.

(1) Differentiated Services Code Point Packet (DSCP) Marking. As part of the session setup process, the LSC controls what DSCP to use in the subsequent session media stream packets. The exact DSCP method used shall comply with Section 5.3.3.3.2. The SUT met all DSCP Packet Marking requirements for IP version 4

(IPv4). The SUT was only tested for IPv6 intra-enclave (see paragraph 11.2. p).

(2) Voice Features and Capabilities. The LSC must provide all of the features listed in Table 5.3.2.2-1 of the UCR. The SUT met all Voice Features and Capabilities requirements.

(3) Public Safety Features. The LSC must provide basic emergency service (911), tracing of terminating calls, outgoing call tracing, and tracing of a call in progress. The SUT met all Public Safety Features requirements.

(4) Assured Services Admission Control – Open Loop. The LSC must meet the Assured Services Admission Control (ASAC) requirements for the LSC and the MFSS. In the execution of ASAC, certain procedures need to be followed, such as (a) actions to be taken if a precedence session request cannot be completed because existing sessions are at equal or higher precedence, or (b) tones to be generated when a session is preempted. The SUT met all ASAC requirements, with the following minor exceptions: IAD phones play T120 instead of Preemption Notification Tone (PNT) during unanswered preemption scenarios. The SUT IAD PNT does not meet the UCR. This was adjudicated as having a minor operational impact.

(5) Signaling Protocols. The LSC must use appropriate signaling for specific trunk types. The control/management protocol between the PEI and the LSC is, in general, proprietary. The control/management protocol between the AEI and the LSC is AS-SIP as specified in Section 5.3.4, AS-SIP Requirements, of this document. The signaling protocol used on UC IP trunks is AS-SIP as specified in Section 5.3.4, AS-SIP Requirements. The LSC and the MG within the MFSS use ANSI T1.619a PRI signaling on DSN PRI trunks. The SUT met all Signaling Protocol requirements. The conditional requirement for Channel Associated Signaling (CAS) was not tested.

(6) Signaling Performance. The LSC has conditional requirements for call setup and tear-down times. The SUT met all signaling performance requirements.

**b. Registration, Authentication, and Failover.**

(1) Registration. Registration and authentication between the LSC and End Instruments (EIs) shall follow the requirements set forth in UCR 2008, Section 5.4, Information Assurance Requirements. This feature is tested by IA and is covered in the IA report (see paragraph 11.3).

(2) Failover. The LSCs shall be registered to a primary and backup MFSS. In case of failure of the primary Multifunction Softswitch (MFSS), the LSC will default to the backup MFSS. The SUT met all failover requirements.

**c. Product Physical, Quality, and Environmental Factors.**

(1) Availability. The Assured Services subsystem shall have a hardware/software availability of 0.99999 (non-availability of no more than 5 minutes per year). This requirement was met via vendor LoC.

(2) Maximum Downtimes. The performance parameters associated with the Assured Service Local Rea network (ASLAN), MFSS, and LSC, when combined, shall meet the following maximum downtime requirements:

- IP (10/100 Ethernet) network links – 35 minutes/year
- IP subscriber – 12 minutes/year

This requirement was met via vendor LoC.

(3) Loss of Packets. For Voice over IP (VoIP) devices, the voice quality shall have a Mean Opinion Score (MOS) of 4.0 (R-Factor equals 80) or better, as measured in accordance with the E-Model. Additionally, these devices shall not lose two or more consecutive packets in a minute and shall not lose more than seven voice packets (excluding signaling packets) in a 5-minute period. The SUT met all Packet Loss requirements for PEIs.

**d. Voice End Instrument.** The SUT met PEI requirements except for the noted discrepancies in the following sub-paragraphs. The SUT was not tested with an AEI because no AEIs have been submitted through the UC process. The AEI requirement was a new requirement for which the 18-month rule (July 2011) applies.

(1) Tones and announcements. Tones and announcements, as required in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, and Section 5.2.2.1.3, Announcements, shall be supported, except for the loss of Command and Control (C2) announcement. The SUT met all requirements for tones and announcements, with the following minor exceptions: The precedence above ROUTINE ring cadence and the IAD PNT do not meet the UCR specifications. The SUT IAD phones play T120 instead of PNT during unanswered call preemptions. These findings have been adjudicated by DISA to have a minor operational impact.

(2) Audio codecs. The LSC shall support the origination and termination of a voice session using the following codecs: G.711 (a-law and µ-law), G.723.1, G.729 or G.729A, and G.722.1. The SUT met all audio codecs requirements for PEIs.

(3) VoIP PEI or AEI Audio Performance Requirements. VoIP PEIs or AEIs (i.e., handset, headset, and hands-free types) shall comply with Telecommunications Industry Association (TIA) -810-B, 3 November 2006. The SUT met all audio performance requirements for PEIs.

(4) VoIP Sampling Standard. For Fixed-to-Fixed calls, the product shall use 20 milliseconds (ms) as the default voice sample length, and as the basis for the voice payload packet size. The SUT met the VoIP sampling standard requirements for PEIs.

(5) Authentication to LSC. The PEI or AEI shall be capable of authenticating itself to its associated LSC and vice versa. The SUT met all PEI to LSC authentication requirements. The SUT was not tested with an AEI. The AEI requirement was a new requirement for which the 18-month rule (July 2011) applies.

(6) Analog Telephone Support. Analog instruments, including secure analog EIs, analog facsimile EIs, and analog modem EIs, shall be supported by the LSC either by a Terminal Adapter or an IAD connected to an Ethernet port. The SUT met all analog telephone support requirements via an IAD.

(7) Softphones. The softphone shall be conceptually identical to a traditional IP “hard” telephone and is required to provide voice features and functionality provided by a traditional IP hard telephone. The SUT met all softphone requirements.

(8) ISDN Basic Rate Interface. The ISDN BRI EIs, including secure ISDN BRI EIs, may be supported by the LSC. This is a conditional requirement. No BRI EIs were provided on the SUT at the time of test. Therefore, ISDN BRI requirements were not tested.

**e. Video End Instruments.** The SUT’s PEIs and AEIs must support both voice and video. The SUT’s PEI support voice only and did not support video. This discrepancy was adjudicated to have a minor operational impact because LSC managed video represents a new requirement not fielded under legacy certifications. The SUT was not tested with an AEI. The AEI requirement was a new requirement for which the 18-month rule (July 2011) applies.

(1) Video End Instrument. Video EIs are considered associated with the LSC and must have been designed in conjunction with the LSC design. An IP video instrument shall be designed in accordance with the acquiring activity requirements. The SUT provided Video End Instruments on the personal computer (PC) Clients only. The video end instruments were tested intra-enclave and inter-enclave to the Avaya MFSS only. In this configuration, the SUT met all video end instrument requirements for the PC Clients. The PEIs did not provide video capabilities. The DISA adjudicated this discrepancy as having a minor operational impact.

(2) Display Messages, Tones, and Announcements. Tones and announcements, as appropriate for voice and video over IP, and as required, in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, 5.2.2.1.3, Announcements, shall be supported by the PEI and AEI. The SUT met all messages, tones, and announcements requirements for the PC Clients. The PEIs did not provide video capabilities. The DISA adjudicated this discrepancy as having a minor operational impact.

(3) Video Codecs (Including Associated Audio Codecs). The product shall support the origination, maintenance, and termination of a video session using the following codecs: one G.xxx and one H.xxx must be used to create and sustain a video

session. The SUT met all video codecs requirements for the PC Clients. The PEIs did not provide video capabilities. The DISA adjudicated this discrepancy as having a minor operational impact.

**f. LSC Requirements.**

(1) PBAS/ASAC Requirements. The LSC shall meet all the requirements for Policy Based Assured Services (PBAS)/ASAC, as appropriate for VoIP and Video over IP services, as specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption. The SUT met all PBAS/ASAC Requirements.

(2) Calling Number Delivery Requirements. The calling number provided to the called party shall be determined by the dial plan serving the calling instrument in accordance with Telcordia Technologies General Requirements (GR)-31-CORE *"CLASS℠ Feature: Calling Number Delivery,"* Issue 1, June 2000. The SUT met all calling number delivery requirements.

(3) LSC Signaling Requirements. The LSC must provide signaling on the line side for local intra-enclave subscriber-to-subscriber calls, and trunk-side signaling for calls between an external enclave and a local subscriber. The SUT met all LSC Signaling Requirements.

(4) Service Requirements under Total Loss of Wide Area Network (WAN) Transport. In the event that a total loss of connectivity to the DISN WAN occurs, the LSC shall provide the following functions:

- Completion of local (intra-enclave) calls.
- Routing of calls to the Public Switched Telephone Network (PSTN) using a local MG (PRI or CAS as required by the local interface).
- User look-up of local directory information.

The SUT met all Service Requirements under Total Loss of WAN Transport.

(5) Local Location Server and Directory. The purpose of the Local Location Server (LLS) is to provide information on call routing and called address translation (where a called address is contained within the called SIP Uniform Resource Identifier (URI) in the form of the called number). The SUT met all LLS and directory requirements.

(6) LSC Transport Interface Functions. The LSC Transport Interface functions provide interface and connectivity functions with the ASLAN and its IP packet transport network. The SUT met all transport interface function requirements.

(7) LSC to PEI, AEI, and Operator Console Status Verification. Periodically, the LSC shall verify the status of its registered and authenticated IP EIs. The SUT met

all status verification requirements for PEIs. The SUT was not tested with AEIs and Attendant Console. Both are new UCR requirements for which the 18-month rule (July 2011) applies.

(8) Line-Side Custom Features Interference. Vendors may implement unique custom features applicable to the line side of the LSC. Line-side custom features must not interfere with the Assured Services requirements. However, JITC did not test any of those features; therefore, they are not certified for use.

(9) Loop Avoidance. During the call establishment process, the product shall be capable of preventing or detecting and stopping hair-pin routing loops over ANSI T1.619a and commercial PRI trunk groups (i.e., T1 PRI and E1 PRI) between a legacy switch (e.g., End Office) and an LSC. The SUT met all Loop Avoidance requirements for T1 PRI; however, E1 PRI was not tested.

**g. Call Connection Agent Requirements.**

(1) CCA IWF Component. The role of the Interworking Function (IWF) within the Call Connection Agent (CCA) is to support all the VoIP and TDM signaling protocols that the LSC supports for EIs, Media Gateways (MGs), and Edge Boundary Controllers (EBCs), and to interwork all these various signaling protocols with one another. The SUT met all CCA IWF requirements for T1 PRI ((ANSI T1.619a and NI-2). Other conditional IWFs (E1 PRI, CAS and Common Channel Signaling 7 (CCS7)) were not tested.

(2) CCA MGC Component. The MG Controller (MGC) within the CCA must control all MGs within the LSC or MFSS, control all trunks within each MG, control all signaling and media streams on each trunk within each MG, accept IP-encapsulated signaling streams from an Signaling Gateway (SG) or MG, and to use either ITU-T recommendation H.248 or a supplier-proprietary protocol to accomplish these controls. The SUT met all CCA MGC requirements.

(3) SG Component. The role of the CCA with respect to the SG is to control all SGs within the network appliance, and to control all signaling links (DoD CCS7) within each SG. The SG is conditional for an LSC and was not tested on the SUT.

(4) CCA-IWF Support for AS-SIP. The CCA IWF shall support the AS-SIP protocol consistent with the detailed AS-SIP protocol requirements in Section 5.3.4, AS-SIP Requirements. The SUT met all requirements for CCA-IWF support for AS-SIP for required T1 PRI (ANSI T1.619a and NI-2) interfaces. Conditional interfaces (E1 PRI, T1 CAS, and T1 CCS7) were not tested.

(5) CCA-IWF Support for SS7. CCA-IWF support for Signaling System 7 (SS7 aka CCS7) is a conditional requirement for LSCs and was not tested on the SUT.

(6) CCA-IWF Support for PRI, via MG. The CCA IWF shall support the United States (US)/National ISDN version of the ISDN PRI protocol. The SUT met all requirements for CCA-IWF support for T1 PRI (ANSI T1.619a and NI-2).

(7) CCA-IWF Support for CAS Trunks via MG. Support for CAS is a conditional requirement for LSCs and was not tested on the SUT.

(8) CCA-IWF Support for PEI and AEI Signaling Protocols. The CCA IWF shall support supplier-proprietary Voice and Video EIs and their associated proprietary EI signaling protocols. The SUT met all requirements for CCA-IWF Support for PEI Signaling Protocols. No AEIs were tested.

(9) CCA-IWF Support for VoIP and TDM Protocol Interworking. The role of the IWF within the CCA is to support all the VoIP and Time Division Multiplexing (TDM) signaling protocols that the appliance supports for PEIs, AEIs, MGs, and EBCs, and interwork all these various signaling protocols with one another. The SUT met all requirements for CCA-IWF support for VoIP and TDM Protocol Interworking required T1 PRI (ANSI T1.619a and NI-2) interfaces. Conditional interfaces (E1 PRI, T1 CAS, and T1 CCS7) were not tested.

(10) CCA Preservation of Call Ringing State during Failure Conditions. The CCA in the LSC, MFSS, and WAN Softswitch (SS) shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within the CCA. This requirement was not tested. This is a new UCR requirement for which the 18-month rule (July 2011) applies.

(11) CCA Interactions with Transport Interface Functions. The CCA interacts with Transport Interface functions by using them to communicate with PEIs, AEIs, the EBC, the MGs, and the SG over the ASLAN. The SUT met all requirements for CCA interactions with Transport Interface Functions exception of AEIs. No AEIs were tested.

(12) CCA Interactions with the EBC. The CCA interacts with the EBC by directing AS-SIP signaling packets to it (for signaling messages destined for an MFSS) and by accepting AS-SIP signaling packets from it (for signaling messages directed to the LSC from an MFSS). The SUT met all requirements for CCA interactions with the EBC.

(13) CCA Support for Admission Control. The CCA interacts with the ASAC component of the LSC and MFSS to perform specific functions related to ASAC, such as counting internal, outgoing, and incoming calls; managing separate call budgets for VoIP and Video over IP calls; and providing preemption. The SUT met all requirements for CCA support for Admission Control

(14) CCA Support for UFS. The User Features and Services (UFS) Server is

responsible for providing features and services to VoIP and Video PEIs/AEIs on an LSC or MFSS, where the CCA alone cannot provide the feature or service. The SUT met all requirements for CCA Support for UFS for PEIs.

(15) CCA Support for IA. The Information Assurance function within the appliance ensures that end users, PEIs, AEIs, MGs, SGs, and EBCs that use the appliance are all properly authenticated and authorized by the appliance. The Information Assurance function ensures that Voice and Video signaling streams that traverse the appliance and its ASLAN are encrypted properly SIP/TLS. The IA requirements are tested separately; see paragraph 11.3.

(16) CCA Interaction with EIs. The LLS provides information on called address translation in response to call routing queries from the CCA. The CCA sends call routing queries to the LLS for both outgoing calls from appliance PEIs or AEIs (i.e., LSC and MFSS) and incoming calls to appliance PEIs or AEIs (i.e., LSC and MFSS). The SUT met all requirements for CCA interaction with PEIs.

(17) CCA Support for AS Voice and Video. The CCA in the MFSS or LSC needs to interact with VoIP PEIs and AEIs served by that MFSS or LSC. The VoIP interface between the PEI and the MFSS or LSC is left up to the network appliance supplier. The VoIP interface between the AEI and the MFSS or LSC is AS-SIP. The SUT met all requirements for CCA support for AS Voice and Video.

(18) CCA Interactions with Service Control Functions. The CCA shall support the ability to remove VoIP and Video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions. The SUT met all requirements for CCA Interactions with Service Control Functions.

(19) CCA Interworking between AS-SIP and CCS7. Interworking is performed at a node with CCA (SIP/CCS7 IWF) functionality that processes/interworks incoming CCS7 messages to outgoing AS-SIP messages, and similarly, incoming AS-SIP messages to outgoing CCS7 messages. This is a conditional requirement for LSCs and is not supported by the SUT and was therefore not tested. CCS7 is provided by the SUTs CS2100 XA-Core SE09.1 MFS.

**h. MG Requirements.**

(1) Role of MG in LSC. The MG supports interconnection of VoIP, Fax over IP (FoIP), and Modem over IP (MoIP) media streams with the LSC media server, which provides tones and announcements for LSC calls and LSC features. To support inter-enclave MoIP and FoIP, the LSC must meet ITU-T V.150.1 requirements. The V.150.1 requirement is a new UCR requirement for which the 18-month (July 2011) rule applies.

(2) MG Support for ASAC. The MG assists the CCA in performing ASAC (i.e., call preemption based on per-call precedence levels) for outgoing TDM calls at

MGs and for incoming TDM calls at MGs. The SUT met all requirements for MG Support for ASAC.

(3) MG and IA Functions. The Information Assurance function within the appliance ensures that end users, PEIs, AEIs, MGs, SGs, and EBCs that use the appliance are all properly authenticated by the appliance. The IA function also ensures that VoIP signaling streams and media streams that traverse the appliance and its ASLAN are properly encrypted, using SIP/TLS and SRTP, respectively. The IA requirements are tested separately; see IA report.

(4) MG Interaction with Service Control Function. The MG is responsible for routing individual VoIP, FoIP, and MoIP media streams to the media server when instructed to do so by the CCA/MGC. When instructed to do so by the CCA/MGC, the MG is responsible for removing individual VoIP, FoIP, and MoIP media streams from the media server, and for either disconnecting them entirely, or routing them on to other LSC end users (e.g., VoIP or video EIs). The SUT met all requirements for MG Interaction with Service Control Function except for V.150.1. The V.150.1 requirement is a new UCR requirement for which the 18-month (July 2011) rule applies.

(5) MG Interactions with IP Transport Interface Functions. The Transport Interface functions in the LSC provide interface and connectivity functions with the ASLAN and its IP packet transport network. The SUT met all requirements for MG Interactions with IP Transport Interface Functions.

(6) MG-EBC interactions. The MG interacts with the EBC by sending SRTP media streams to it (for call media destined for a PEI, AEI, or MG that is served by another appliance outside the LSC), or by accepting SRTP media streams from it (for call media arriving from a PEI, AEI, or MG that is served by another appliance outside the LSC). The SUT met all requirements for MG-EBC interactions PEI interactions.

(7) MG IP-Based PSTN Interface Requirements. Voice and Video over IP interfaces from the UC network to the PSTN have not been defined. Interfaces from an LSC or MFSS to the PSTN will be via an MG with TDM interfaces as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features.

(8) MG Interaction with VoIP EIs. The MG in the MFSS or LSC needs to interact with VoIP EIs served by that MFSS or LSC, and with VoIP EIs served by other MFSSs or LSCs. The VoIP signaling interface between the PEI and the MFSS or LSC is left up to the network appliance supplier. The VoIP signaling interface between the AEI and the MFSS or LSC is AS-SIP. The SUT met all MG Interaction with VoIP EIs requirements.

(9) MG Support for User Features and Services. The MG shall support the operation of features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches today. The SUT met all requirements for MG Support for User Features and Services.

(10) MG Interface to TDM network elements in DoD Networks. Each appliance MG shall support TDM trunk groups that can interconnect with the following devices in DoD networks, in the US and worldwide: Private Branch Exchanges (PBXs), Small End Offices (SMEOs), EOs, and MFSS. The SUT met all requirements for MG Interface to TDM devices in DoD Networks.

(11) MG Interface to TDM Allied and Coalition. The appliance suppliers should support TDM trunk groups on their MG product that can interconnect with devices in US allied and coalition partner networks worldwide. This requirement is conditional and was not tested on the SUT.

(12) MG Interface to TDM PSTN in US. Each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the United States. The SUT met all requirements for MG Interface to TDM PSTN in the US using T1 PRI (NI-2).

(13) MG Interfaces to TDM PSTN OCONUS. The appliance supplier (i.e., LSC or MFSS supplier) should support TDM trunk groups on its MG product that can interconnect with devices in foreign country telephone networks (OCONUS) worldwide. This requirement was not tested on the SUT.

(14) MG Support for CCS7. The MG shall support TDM trunk groups that are controlled by a separate CCA-to-SG signaling link that carries DoD CCS7 protocol. The MG shall support these TDM trunk groups, and the SG shall support DoD CCS7 signaling. This conditional requirement was not tested on the SUT.

(15) MG Support for ISDN PRI Trunks. The MG shall support ISDN PRI trunk groups that carry the US/National ISDN version of the ISDN PRI protocol. The SUT met all requirements for MG Support for ISDN T1 PRI (ANSI T1.619a and NI-2) Trunks.

(16) MG Support for CAS Trunks. The MG shall support CAS trunk groups that carry the US version of the CAS protocol. The CAS is a conditional requirement for LSCs and was not tested on the SUT.

(17) MG requirements for VoIP Internal Interfaces. The MG shall connect to the ASLAN of the appliance using the physical layer and data link layer protocols of the ASLAN. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as a physical layer and data link layer endpoint on a LAN switch in the ASLAN. The SUT met all requirements for VoIP Internal Interfaces for PEIs.

(18) MG Echo Cancellation. The MG shall provide an Echo cancellation (EC) capability with an echo path capacity (echo tail length) of at least 64 ms. The MG shall provide echo cancellation for voice, G3 Fax, and VBD modem fax calls. Each MG EC shall be equipped with an “echo canceller disabling signal” tone detector. The SUT met

all requirements for MG Echo Cancellation.

(19) MG Clock Timing. The MG shall derive its clock timing from a designated T1 or PRI interface. The SUT met all MG Clock Timing requirements.

(20) MG V.150.1. When the MG uses V.150.1 inband signaling to transition between audio, FoIP, modem relay, or variable bit data (VBD) states or modes, the MG shall continue to use the established session's protocol (e.g., decimal 17 for User Datagram Protocol (UDP) and port numbers so that the transition is transparent to the EBC. The V.150.1 requirement is a new UCR requirement for which the 18-month (July 2011) rule applies.

(21) MG Preservation of Call Ringing during Failure. The LSC MG, MFSS MG, and WAN SS MG shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within that MG. This requirement was not tested on the SUT. This is a new UCR requirement for which the 18-month rule (July 2011) applies.

**i. SG Requirements.**

(1) SG and CCS7 network Interactions. The SG shall support signaling connectivity to the DoD CCS7 network based on UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features, specifications for CCS7. This is a conditional requirement for LSCs and was not tested on the SUT.

(2) SG Interactions with CCA. The SG shall support a supplier-specific interface to the CCA for interactions between the SG and CCA. This is a conditional requirement for LSCs and was not tested on the SUT.

(3) SG Interworking Functions. The SG will terminate CCS7 links on its CCS7 side and transport the CCS7 call control and service control protocols (i.e., ISUP and TCAP) to the CCA. Similarly, the SG will receive CCS7 call control and service control messages from the CCA. The SG is responsible for the appropriate formatting of the messages for transmission on the CCS7 links. This is a conditional requirement for LSCs and was not tested on the SUT.

**j. Worldwide Numbering and Dialing Plan Requirements.**

(1) Worldwide Numbering and Dialing Plan. The precedence level and dialed number input to the PEI or AEI shall be as specified in UCR 2008, Section 5.2.3.5.1.2, Interswitch and Intraswitch Dialing. The SUT met all requirements for Worldwide Numbering and Dialing Plan (WWNDP) for PEIs.

(2) DSN Worldwide Numbering and Dialing Plan. The LSCs must support

DSN WWNDP and must support mapping of DSN telephone numbers to SIP URIs, provides examples of DSN numbers using SIP URIs that use the syntax defined in RFC 3966. The SUT met all DSN WWNDP requirements.

**k. Commercial Cost Avoidance (CCA).** The LSC must use a Commercial Cost Avoidance functionality to route calls from an IP EI to a DSN E.164 number in a manner which will minimize commercial costs associated with PSTN calls. This requirement was tested and was partially met by the SUT. The SUT Ethernet interface configured for CCA queries to the RTS Routing Database Server support IPv6 for media but not signaling. The Vendor submitted an IPv6 LoC with noted discrepancies which include the interface for Commercial Cost Avoidance functionality. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.

**l. AS-SIP Based Support for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices).** The LSC shall support all mandatory requirements in RFC 3842. The LSC shall support all mandatory requirements in IETF Internet Draft draft-levy-sip-diversion-08.txt, Diversion Indication in SIP. This is a conditional requirement. The JITC did not test any AS-SIP external devices.

**m. Precedence Call Diversion.** The AS-SIP signaling appliance shall divert all unanswered VoIP calls above the ROUTINE precedence level to a designated directory number (DN) for precedence call diversion. The SUT met all precedence call diversion requirements.

**n. Attendant Station Features.** No attendant station was provided on the SUT at the time of test; therefore, none of the following features were tested. The Attendant Console functionality is a new UCR requirement for which the 18-month rule (July 2011) applies.

(1) Precedence and Preemption. The RTS Attendant Console shall interoperate with PBAS/ASAC.

(2) Call Display. The RTS Attendant Console shall provide a visual display of each precedence level and the calling number, for incoming direct dialed calls to the attendant, and diverted calls to the attendant.

(3) Class of Service Override. If the LSC, MFSS, or WAN SS supports assignment of a Class of Service (CoS) to an individual EI, then this appliance and the attendant console shall give the attendant the ability to override any incoming call's calling party CoS (based on calling area or precedence) on a call-by-call basis.

(4) Busy Override and Busy Verification. The appliance and the attendant console shall give the attendant the ability to verify and override a busy line condition.

(5) Night service. The appliance and the attendant console shall have the ability to route all calls that are normally directed to the console to a separate night service deflection number.

(6) Automatic Recall of Attendant. When an attendant redirects an incoming call to a destination station, and that station is either busy or does not answer the call within a preset time, the appliance and the attendant console shall ensure that calling party on the redirected call is recalled automatically to the console.

(7) Calls in Queue to the Attendant. The appliance and the attendant console shall have the ability to place calls (both directed to the attendant and diverted to the attendant) into a waiting queue.

**o. AS-SIP Requirements.**

(1) SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs. The LSCs that support SIP EIs MUST comply with the differentiated set of requirements defined for SIP EIs if they serve SIP EIs, and LSCs must comply with the differentiated set of requirements defined for H.323 and/or vendor-proprietary EIs if they serve H.323 and/or vendor-proprietary EIs. This requirement was not tested on the SUT. This represents a new feature in the UCR for which the 18-month rule applies. The vendor has until July 2011 to comply with this feature.

(2) SIP Session Keep-Alive Timer. The AS-SIP signaling appliances MUST support the keep-alive mechanism for SIP sessions in accordance with request for comment (RFC) 4028. The SUT met all keep-alive timer requirements.

(3) Session Description Protocol (SDP). A session description consists of a session-level description (details that apply to the whole session and all media streams) and optionally several media-level descriptions (details that apply to a single media stream). The LSC must support SDP in accordance with RFC 2327. The SUT met all SDP requirements.

(4) Precedence and Preemption. The LSC must meet the detailed requirements for the execution of preemption and the handling of precedence information as defined in section 5.3.4.2.10 of the UCR. The SUT met all precedence and preemption requirements.

(5) Video Telephony – General Rules. Video calls must meet the detailed requirements for video telephony messaging as defined in section 5.3.4.12 of the UCR. Video telephony requirements were tested on the SUT using softphone PC Clients. The SUT met all video telephony requirements.

(6) Calling Services. The LSC must meet AS-SIP call flow requirements for calling services features as defined in section 5.3.4.13 of the UCR. The SUT met all

calling services requirements.

(7) SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances. This specification uses SIP translation for converting between ISDN User part (ISUP) signaling and AS-SIP signaling but does not use SIP encapsulation of ISUP. This requirement applies to translations between AS-SIP and CCS7. Because CCS7 is a conditional requirement for LSCs and not supported by the SUT, this requirement was not tested.

(8) Relevant Timers for the Terminating Gateway and the Originating Gateway. This requirement applies to gateways between AS-SIP and CCS7 links. Because CCS7 is a conditional requirement for LSCs and not supported by the SUT, this requirement was not tested.

(9) SIP Requirements for Interworking AS-SIP Signaling Appliances. Interworking AS-SIP signaling appliances must comply with UCR 2008 Section 5.3.4.7.1, AS-SIP Signaling Appliances and AS-SIP EIs, as well as the additional general requirements in UCR 2008 Section 5.3.4.16. The SUT met all requirements for interworking AS-SIP signaling appliances.

(10) Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances. Interworking AS-SIP signaling appliances must comply with UCR 2008, Section 5.3.4.8, SIP Session Keep-Alive Timer, as well as the additional E1 requirements listed in UCR 2008 Section 5.3.4.17. The SUT met all keep-alive timer requirements for interworking AS-SIP signaling appliances.

(11) Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances. The LSC must meet all requirements for header fields as listed in UCR 2008 section 5.3.4.18. The SUT met all requirements for precedence and preemption extensions for interworking AS-SIP signaling appliances.

(12) Supplementary Services. The LSC must meet call flow requirements as described in section 5.3.4.19 for supplementary services. The SUT met all supplementary services requirements.

**p. IPv6 Requirements.**

(1) Product Requirements. This requirement was test intra-enclave, and the vendor has submitted an LoC for IPv6, stating the SUT is not compliant with the following:

- IAD (MP 112/124): Their solution does not support IPv6 (Signaling or Media) with their MP112 and MP124 analog IADs. Per the UCR Change 1 an LSC MG is required to support IPv6 if the packets exit the enclave which they do.

- The SESM Core supports IPv4 only for signaling and both IPv4 and

IPv6 dual stack for inter-enclave. The requirement is IPv6 and IPv4 for both signaling and media.

- The Audio Codes MG3K supports IPv4 only for signaling and both IPv4 and IPv6 dual stack for media intra and inter-enclave. The requirement is IPv6 and IPv4 for both signaling and media.

- IPv6 inter-switch requirements have not been tested by the JITC.

**q. Network Management.**

(1) LSC Management Function. The LSC Management function supports functions for LSC Fault, Configuration, Accounting, Performance, and Security (FCAPS) management and audit logs. The SUT met all requirements for the LSC Management Function via LoC.

(2) VVoIP NMS Interface Requirements. The physical interface between the DISA Voice and Video over IP (VVoIP) element management system (EMS) and the network components (i.e., LSC, MFSS, and EBC) is 10/100-Megabits per second (Mbps) Ethernet interface. The interface will work in either of the two following modes using auto-negotiation: Institute of Electrical and Electronics Engineers, Inc. (IEEE), Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995. The SUT meets all VVoIP NMS Interface Requirements.

(3) General Management Requirements. The LSC components shall each have an individual pair of Ethernet interfaces for management purposes, even in cases where the MFSS or LSC component contains multiple physical devices. The SUT met all general management requirements via LoC.

(4) Requirement for Fault, Configuration, Accounting, Performance, and Security Management. The LSC must meet all general requirements for the FCAPS management functional areas as defined in UCR 2008 Section 5.3.2.17. The SUT does not meet FCAPS requirements. The SUT does not fully comply with the UCR Change 1 section 5.3.2.17.2. The SUT only supports SNMPv2 protected by IPSEC transport, while the UCR requires SNMPv3. The SUT does not download software automatically as required. The SUT can receive software electronically or physically, but does not automatically retrieve, install or rollback software updates. The SUT does not backup the software application during the backup procedure as required and has been adjudicated as minor.

(5) NM requirements of Appliance Functions. The LSC must meet all management requirements for ASAC, CCA, SG, and MG functions as defined in UCR 2008 Section 5.3.2.18. The SUT does not meet the NM requirements of appliance functions. The SUT does not download software automatically. The SUT electronically or physically, but does not automatically retrieve, install or rollback software updates. The SUT AudioCodes EMS is reachable by the Local EMS, but not by the VVoIP EMS.

ADIMSS is not managing the LSC, so this access is provided only to the Local EMS. The UCR requires the EMS to be reachable by the VVoIP EMS. No consolidation of alarms is performed by the SUT to provide only root cause alarms as required. The suppression function is not supported by the SUT as required. The SUT MG keeps history for 2 intervals of 15 minutes, instead of the required current interval and 8 hours of history data. The SUT does not support management requirements for ASAC as defined in the UCR and has been adjudicated as minor.

(6) Accounting Management. Accounting management identifies a set of events during which call detail information is collected. These events are call connect, call attempt, and call disconnect. When these events are detected, specific call data will be provided by the network appliances that were involved in the event. The SUT met all accounting management requirements via LoC.

**11.3 Information Assurance.** The IA report is published in a separate report, Reference (e).

**11.4 Other.** None

**12. TEST AND ANALYSIS REPORT.** No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System 2-7 Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

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## SYSTEM FUNCTIONAL AND CAPABILITY REQUIREMENTS

The local session controllers have required and conditional features and capabilities that are established by the Unified Capabilities Requirements (UCR). The System Under Test (SUT) need not provide conditional requirements. If they are provided, they must function according to the specified requirements. The detailed Functional requirements (FR) and Capability Requirements for Internet Protocol Call Control products (Multi-Function SoftSwitch (MFSS), Local Session Controller (LSC), and Wide Area Network SoftSwitch (WAN SS) are listed in Table 3-1. Detailed Information Assurance (IA) requirements are included in Reference (e) and are not listed below.

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table**

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 1 | As part of the session setup process, the LSC controls what DSCP to use in the subsequent session media stream packets. (two sub requirements) | 5.3.2.2.1.4 | Y | Y | |
| 2 | The SUT must provide the following features: Precedence Call Waiting, Call Forwarding, Call Transfer, Call Hold, Three-Way Calling, Hotline Service, and Calling Party and Called Party ID. | Table 5.3.2.2-1 | Y | Y | |
| 3 | Calls to a DN that does not have any CF feature activated shall be delivered to the DN EI IAW the MLPP procedures specified in UCR 2008, Section 5.2.2 Multilevel Precedence and Preemption | 5.3.2.2.2.1.1 | Y | Y | |
| 4 | Call forwarding, when activated on a line DN, shall allow any terminating call at a ROUTINE DSN precedence level, to be completed to the designated destination (IAW the call forward options activated), and shall comply with the requirements as stated in Telcordia Technologies GR-217-CORE, GR-580-CORE, and GR-586-CORE. | 5.3.2.2.2.1.1 | Y | Y | |
| 5 | Calls to 911 shall be preempted in accordance with assured service priority rules specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption. | 5.3.2.2.2.2.1 | Y | Y | |
| 6 | The Tracing of Terminating Calls feature identifies the calling number on intraoffice and interoffice calls terminating to a specified DN. When this feature is activated, the originating DN, the terminating DN, and the time and date are printed out for each call to the specified line. | 5.3.2.2.2.2.2 | Y | Y | |
| 7 | The Outgoing Call Tracing feature allows the tracing of nuisance calls to a specified DN suspected of originating from a given local office. The tracing is activated when the specified DN is entered. A printout of the originating DN, and the time and date, are generated for every call to the specified DN. | 5.3.2.2.2.2.3 | Y | Y | |
| 8 | The Tracing of a Call in Progress feature identifies the originating DN for a call in progress. Authorized personnel entering a request that includes the specific terminating DN involved in the call activate the feature. | 5.3.2.2.2.2.4 | Y | Y | |
| 9 | The Tandem Call Trace feature identifies the incoming trunk of a tandem call to a specified office DN. The feature is activated by entering the specified distant office DN for a tandem call trace. A printout of the incoming trunk number and terminating DN, and the time and date, is generated for every call to the specified DN. | 5.3.2.2.2.2.5 | Y | Y | |
| 10 | One voice session budget unit shall be equivalent to 110 kilobits per second (kbps) of access circuit bandwidth independent of the PEI or AEI codec used. This includes ITU-T Recommendation G.711 encoding rate plus Internet Protocol Version 6 (IPv6) packet overhead plus ASLAN Ethernet overhead. IPv6 overhead, not IPv4 overhead, is used to determine bandwidth equivalents here. | 5.3.2.2.2.3.1 | Y | Y | |
| 11 | If the MFSS's count of an IPC is greater than or equal to the corresponding IPB, and it receives an INVITE request for a precedence session, the MFSS shall preempt a lower priority session (if such a session exists), and then proceed with processing the higher precedence session connect request. | 5.3.2.2.2.3.1.2 | Y | | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 12 | If the MFSS receives a CCA-ID for which there is no entry in ASAC budget table, the SS will reject the session and generate an alarm for the EMS. | 5.3.2.2.2.3.1.2 | Y | | |
| 13 | If necessary, the MFSS will preempt for a session request that is at precedence level FLASH OVERRIDE or FLASH and the counts equal the budgets. | 5.3.2.2.2.3.2 | Y | | |
| 14 | Registration and authentication between NEs shall follow the requirements set forth in UCR 2008, Section 5.4, Information Assurance Requirements. | 5.3.2.3.1 | Y | Y | |
| 15 | The LSCs shall be registered to a primary and backup MFSS. In case of failure of the primary MFSS, the LSC will default to the backup MFSS. | 5.3.2.3.2 | Y | Y | |
| 16 | The LSC shall send an OPTIONS request with a Request-URI identifying the primary SS (the Request-URI does not have a userinfo part) on a configurable periodic time interval (default equals 45 seconds; minimum time interval equals 35 seconds). (4 sub requirements) | 5.3.2.3.2.2 | | Y | |
| 17 | When a properly functioning primary SS receives the OPTIONS request from a served LSC, the primary SS shall respond with a 200 OK response that includes the Accept header and the Supported header. | 5.3.2.3.2.2 | Y | | Y |
| 18 | When the LSC sends a defined configurable number of successive OPTIONS requests (default equals 2) for which there either is no response or the response is a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response, then it must failover to the secondary SS. (3 sub requirements) | 5.3.2.3.2.3 | | Y | |
| 19 | If the LSC receives a 200 OK response to an OPTIONS request from the primary SS before the configurable number of successive failures to the OPTIONS requests (default equals 2) has been reached, then no action is taken to failover to the secondary SS. | 5.3.2.3.2.3 | | Y | |
| 20 | Upon failover, the LSC will send OPTIONS requests to the primary SS at a failback configurable periodic time interval (default equals 60 seconds; minimum time interval equals 35 seconds). (4 sub requirements) | 5.3.2.3.2.4 | | Y | |
| 21 | Each SS shall send an OPTIONS request to every other SS on a “standard” configurable periodic time interval (default equals 45 seconds; minimum time interval equals 35 seconds). | 5.3.2.3.2.5 | Y | | Y |
| 22 | Whenever an originating SS sends an INVITE request to another SS and receives either a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response and the originating SS is not already awaiting a response to a pending OPTIONS request to the other SS, then the originating SS shall send an OPTIONS request with a Request-URI identifying the SS. | 5.3.2.3.2.5 | Y | | Y |
| 23 | When a properly functioning SS receives the OPTIONS request, the SS shall respond with a 200 OK response that includes the Accept header and the Supported header. | 5.3.2.3.2.5 | Y | | Y |
| 24 | Each MFSS (SS) shall be configured with knowledge of each pair of SSs that act as backups for each other. (7 sub requirements) | 5.3.2.3.2.6 | Y | | Y |
| 25 | Upon failover, the SS will send OPTIONS requests to the failed SS at a “failback” configurable periodic time interval (default equals 60 seconds; minimum time interval equals 35 seconds). (5 sub requirements) | 5.3.2.3.2.7 | Y | | Y |
| 26 | The Assured Services subsystem shall have a hardware/software availability of 0.99999 (nonavailability of no more than 5 minutes per year). | 5.3.2.5.2.1 | Y | Y | |
| 27 | The performance parameters associated with the ASLAN, MFSS, and LSC, when combined, shall meet the following maximum downtime requirements:<br>• IP (10/100 Ethernet) network links – 35 minutes/year<br>• IP subscriber – 12 minutes/year | 5.3.2.5.2.2 | Y | Y | |
| 28 | For these VoIP devices, the voice quality shall have a MOS of 4.0 (R-Factor equals 80) or better, as measured in accordance with the E-Model. Additionally, these devices shall not lose two or more consecutive packets in a minute and shall not lose more than seven voice packets (excluding signaling packets) in a 5-minute period. | 5.3.2.5.4 | Y | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 29 | An IP voice instrument shall be designed in accordance with the acquiring activity requirements, but the following capabilities are specifically required as indicated:<br>• **[Objective]** DoD Common Access Card (CAC) reader<br>• **[Required]** Display calling number<br>• **[Required]** Display precedence level of the session<br>• **[Required]** Support for Dynamic Host Configuration Protocol (DHCP). | 5.3.2.6.1 | Y | Y | |
| 30 | Tones and announcements, as required in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, and Section 5.2.2.1.3, Announcements, shall be supported, except for the loss of C2 announcement. | 5.3.2.6.1.1 | Y | Y | |
| 31 | The product shall support the origination and termination of a voice session using the following codecs:<br>• ITU-T Recommendation G.711, to include both the µ-law and A-law algorithms<br>• ITU-T Recommendation G.723.1<br>• ITU-T Recommendation G.729 or G.729A<br>• ITU-T Recommendation G.722.1 | 5.3.2.6.1.2 | Y | Y | |
| 32 | Voice over IP PEIs or AEIs (i.e., handset, headset, and hands-free types) shall comply with TIA-810-B, November 3, 2006. | 5.3.2.6.1.3 | Y | Y | |
| 33 | For Fixed-to-Fixed calls, the product shall use 20 ms as the default voice sample length, and as the basis for the voice payload packet size. | 5.3.2.6.1.4 | Y | Y | |
| 34 | The PEI or AEI shall be capable of authenticating itself to its associated LSC and vice versa. | 5.3.2.6.1.5<br>5.3.2.6.2.3 | Y | Y | |
| 35 | Analog instruments, including secure analog EIs, analog facsimile EIs, and analog modem EIs, shall be supported by the LSC either by a TA or an Integrated Access Device (IAD) connected to an Ethernet port. | 5.3.2.6.1.6 | Y | Y | |
| 36 | The LSC shall meet all the requirements for PBAS/ASAC, as appropriate for VoIP and Video over IP services, as specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption. | 5.3.2.7.2.1 | | Y | |
| 37 | The LSC shall support CND, as specified in UCR 2008, Section 5.2.3.5.1.8.2, Calling Number Delivery. | 5.3.2.7.2.2 | | Y | |
| 38 | The LSC must provide signaling on the line side for local intra-enclave subscriber-to-subscriber calls, and trunk-side signaling for calls between an external enclave and a local subscriber. | 5.3.2.7.2.3 | | Y | |
| 39 | In the event that a total loss of connectivity to the DISN WAN occurs, the LSC shall provide the following functions:<br>• Completion of local (intra-enclave) calls<br>• Routing of calls to the PSTN using a local MG (PRI or CAS as required by the local interface)<br>• User look-up of local directory information | 5.3.2.7.2.4 | | Y | |
| 40 | The LSC Management function supports functions for LSC FCAPS management and audit logs. Collectively, these functions are called FCAPS Management and Audit Logs. | 5.3.2.7.2.6 | | Y | |
| 41 | The LSC Transport Interface functions provide interface and connectivity functions with the ASLAN and its IP packet transport network. | 5.3.2.7.2.7 | | Y | |
| 42 | The LSC shall provide an interface to the DISA NMS. The interface consists of a 10/100-Mbps Ethernet connection | 5.3.2.7.2.8 | | Y | |
| 43 | Periodically, the LSC shall verify the status of its registered and authenticated IP EIs, including operator (dial service attendant) consoles. The verification interval shall be configurable with the default set at 5 minutes. | 5.3.2.7.2.10 | | Y | |
| 44 | Line-side custom features must not interfere with the Assured Services requirements. | 5.3.2.7.2.11 | | Y | |
| 45 | During the call establishment process, the product shall be capable of preventing or detecting and stopping hair-pin routing loops over ANSI T1.619a and commercial PRI trunk groups (i.e., T1 PRI and E1 PRI) between a legacy switch (e.g., TDM EO) and an LSC | 5.3.2.7.3 | | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 46 | When the AS-SIP TDM Gateway receives a call request over an ISDN MLPP PRI then the AS-SIP TDM Gateway MUST map the telephony numbers received from the Q.931 SETUP message to SIP URIs | 5.3.2.7.4.3.3 | Y | Y | |
| 47 | The AS-SIP TDM Gateway MG MUST support the ITU-T Recommendation G.711 (µ-law and A-law) audio codec. | 5.3.2.7.4.3.4 | Y | Y | |
| 48 | The AS-SIP TDM Gateway MG MUST support RFC 4040 and the AS-SIP TDM Gateway MUST support the signaling for establishing the 64kbps unrestricted bearer per Section 5.3.4.7.7, 64 kbps Transparent Calls (Clear Channel). | 5.3.2.7.4.3.4 | Y | Y | |
| 49 | The AS-SIP TDM Gateway MG MUST support T.38 Fax Relay | 5.3.2.7.4.3.4 | Y | Y | |
| 50 | The AS-SIP TDM Gateway MG MUST support the SCIP-216 subset of V.150.1 Modem Relay (see Section 5.3.2.21.2, RTS SCIP Gateway Requirements) and the AS-SIP TDM Gateway MUST support the AS-SIP signaling requirements in support of modem relay | 5.3.2.7.4.3.4 | Y | Y | |
| 51 | The AS-SIP TDM Gateway MUST satisfy the Information Assurance requirements in Section 5.4 Information Assurance for a media gateway. This requirement is covered under a separate IA certification letter. | 5.3.2.7.4.3.5 | Y | Y | |
| 52 | The AS-SIP TDM Gateway MUST provide an interface to the DISA NMS. The interface MUST consist of a 10/100-Mbps Ethernet connection | 5.3.2.7.4.3.9 | Y | Y | |
| 53 | The AS-SIP IP Gateway MUST implement call count thresholds for voice sessions and for video sessions in order to perform Session Admission Control (SAC). | 5.3.2.7.5.1.1 | Y | Y | |
| 54 | The requirements for the TDM side of the MFSS are entirely the same as for the DSN MFS specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features. The TDM side of the MFSS must meet these requirements. | 5.3.2.8.2.1 | Y | | |
| 55 | MFSS shall support PRI signaling for TDM communication with other systems. | 5.3.2.8.2.3 | Y | | |
| 56 | The TDM side of the MFSS shall support CCS7 signaling for communication with other TDM systems. | 5.3.2.8.2.3 | Y | | |
| 57 | MFSS shall support AS-SIP signaling for IP communication with other MFSSs and LSCs. | 5.3.2.8.2.3 | Y | | |
| 58 | The MFSS shall provide internal signaling and media conversion for calls between the TDM side and SS side of the MFSS. | 5.3.2.8.2.3 | Y | | |
| 59 | The CCA/SG/MGC/MG complex in the SS side of the MFSS needs to interface and interact with the EO and Tandem functions in the TDM side of the MFSS. | 5.3.2.8.2.4 | Y | | |
| 60 | The MFSS MG must support internal MG connections that interconnect the SS side of the MFSS with the EO and Tandem functions on the TDM side of the MFSS. | 5.3.2.8.2.4 | Y | | |
| 61 | The MFSS MG shall interact with the MFSS MGC so that Internal MG connections between the SS and TDM sides of the MFSS support<br>(1) Intra-MFSS calls between TDM EIs connected to the TDM side, and PEIs/AEIs connected to the SS side of the MFSS<br>(2) Incoming and outgoing calls to/from systems external to the MFSS that require conversion between TDM and IP | 5.3.2.8.2.4 | Y | | |
| 62 | When a U.S. ISDN PRI-based connection is used between the SS and TDM sides of the MFSS, the MFSS MG shall interact with the MFSS MGC so that U.S. ISDN PRI signaling (National ISDN PRI signaling with the Precedence Level IE and related MLPP IEs included) is used between the softswitch and TDM sides, and the T1.619/T1.619a version of the ISDN PRI MLPP feature operates correctly between the SS and TDM sides of the MFSS, for both VoIP-to-TDM calls and TDM-to-VoIP calls over this trunk group. | 5.3.2.8.2.4 | Y | | |
| 63 | The SS side of the MFSS shall meet all the requirements for MLPP, as appropriate for VoIP and Video over IP services, as specified in Section 5.2.2, Multilevel Precedence and Preemption. | 5.3.2.8.2.6 | Y | | |
| 64 | The SS side of the MFSS shall support CND as specified in UCR 2008, Section 5.2.3.5.1.8.2, Calling Number Delivery. | 5.3.2.8.2.6 | Y | | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 65 | The requirements for SCS functions (i.e., CCA, IWF, MG, MGC, and SG) and NM are provided in separate sections of this document. The MFSS must meet all these requirements. | 5.3.2.8.2.6 | Y | | |
| 66 | The CCA IWF must support AS-SIP and ISDN PRI protocols. | 5.3.2.9.2.1 | Y | Y | |
| 67 | The MGC within the CCA must control all MGs within the LSC or MFSS, support DoD ISDN trunks, control all signaling and media streams on each trunk within each MG, and accept IP-encapsulated singaling streams from an SG or MG. | 5.3.2.9.2.2 | Y | Y | |
| 68 | The CCA shall be responsible for controlling all the SGs within the MFSS and LSC. | 5.3.2.9.2.3 | Y | C | |
| 69 | The CCA shall be responsible for controlling each signaling link within each SG within the MFSS or LSC. | 5.3.2.9.2.3 | Y | C | |
| 70 | The CCA shall be responsible for controlling the DoD CCS7 signaling stream(s) within each signaling link within each SG. | 5.3.2.9.2.3 | Y | C | |
| 71 | Within the network appliance (i.e., MFSS and LSC), the CCA shall use either an IETF-standard set of CCS7-over-IP protocols, or a supplier-proprietary protocol to accomplish the above SG, signaling link, and signaling stream controls. | 5.3.2.9.2.3 | Y | C | |
| 72 | The CCA IWF shall support the AS-SIP protocol consistent with the detailed AS-SIP protocol requirements in Section 5.3.4, AS-SIP Requirements. | 5.3.2.9.5.1 | Y | Y | |
| 73 | The CCA IWF shall use the AS-SIP protocol on LSC-MFSS and MFSS-MFSS sessions. | 5.3.2.9.5.1 | Y | Y | |
| 74 | When the CCA IWF uses the AS-SIP protocol over the Access Segment between the EBC and the DISN WAN, or over the DISN WAN itself, the CCA IWF shall secure the AS-SIP protocol using TLS. | 5.3.2.9.5.1 | Y | Y | |
| 75 | The CCA IWF shall support the U.S./National ISDN version of the ISDN PRI protocol. | 5.3.2.9.5.3 | Y | Y | |
| 76 | The CCA IWF shall support reception of ISDN PRI messages from the MG and transmission of ISDN PRI messages to the MG. | 5.3.2.9.5.3 | Y | Y | |
| 77 | The CCA IWF shall be able to determine the ISDN PRI (and its D-Channel signaling link) that an incoming PRI message was received on, when processing an incoming PRI message from the MG. | 5.3.2.9.5.3 | Y | Y | |
| 78 | The CCA IWF shall be able to identify the ISDN PRI (and its D-Channel signaling link) that an outgoing PRI message will be sent on, when generating an outgoing PRI message to the MG. | 5.3.2.9.5.3 | Y | Y | |
| 79 | The CCA IWF shall be able to support multiple ISDN PRIs (and their D-Channel signaling links) at the MG, where each PRI is connected to a different PRI end point. | 5.3.2.9.5.3 | Y | Y | |
| 80 | The CCA IWF shall be able to differentiate between the individual ISDN PRIs (and their D-Channel signaling links) at the MG. | 5.3.2.9.5.3 | Y | Y | |
| 81 | The CCA IWF shall support the full set of ISDN MLPP requirements in ANSI T1.619 and ANSI T1.619a. | 5.3.2.9.5.3 | Y | Y | |
| 82 | The CCA IWF shall not support any of the ISDN MLPP requirements in ANSI T1.619 and ANSI T1.619a, on ISDN PRIs to TDM PBXs and switches in the U.S. PSTN. | 5.3.2.9.5.3 | Y | Y | |
| 83 | On ISDN PRIs from the CCA/MG to TDM PBXs and switches in allied and coalition partners (where those networks support U.S. "National ISDN" PRI), the CCA IWF shall support a DoD-user-configurable per-PRI option that allows the PRI to support or not support the ANSI T1.619/619a PRI MLPP feature on calls to and from that PRI. | 5.3.2.9.5.3 | Y | Y | |
| 84 | The CCA IWF shall be able to associate individual PRI configuration data with each individual PRI served by the MG and the CCA. The CCA IWF shall not require groups of PRIs served by the MG and the CCA to share "common" PRI configuration data. | 5.3.2.9.5.3 | Y | Y | |
| 85 | The CCA IWF shall support supplier-proprietary Voice and Video EIs and their associated proprietary EI signaling protocols. | 5.3.2.9.5.5 | Y | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 86 | The CCA in the LSC, MFSS, and WAN SS shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within the CCA. | 5.3.2.9.6 | Y | Y | |
| 87 | The MFSS CCA shall be able to support MG connections between the SS side of the MFSS and the EO and Tandem functions on the TDM side of the MFSS. | 5.3.2.10.1 | Y | | |
| 88 | The CCA shall support assignment of the following items to itself:<br>• Only one CCA IP address (this one IP address may be implemented in the CCA as either a single logical IP address or a single physical IP address),<br>• A CCA Fully Qualified Domain Name (FQDN) that maps to that IP address, and<br>• A CCA SIP URI that uses that CCA FQDN as its domain name, and maps to the “SIP B2BUA” function within the CCA itself. | 5.3.2.10.3 | Y | Y | |
| 89 | The CCA shall support assignment of the following items to each SIP and AS-SIP PEI and AEI on the Appliance LAN:<br>• Only one PEI or AEI IP address,<br>• A PEI or AEI FQDN that maps to that IP address, and<br>• A PEI or AEI SIP URI that uses that PEI or AEI FQDN as its domain name, and maps to the “SIP User Agent” function within the PEI or AEI. | 5.3.2.10.3 | Y | Y | |
| 90 | The CCA shall support assignment of the following items to each MG on the Appliance LAN:<br>• Only one MG IP address (this one IP address may be implemented in the MG as either a single logical IP address or a single physical IP address),<br>• An MG FQDN that maps to that IP address, and<br>• An MG SIP URI that uses that MG FQDN as its domain name, and maps to the “UC Signaling and Media End Point” function within the MG. | 5.3.2.10.3 | Y | Y | |
| 91 | The CCA shall support assignment of the following items to each SG on the Appliance LAN:<br>• Only one SG IP address (this one IP address may be implemented in the SG as either a single logical IP address or a single physical IP address),<br>• An SG FQDN that maps to that IP address, and<br>• An SG SIP URI that uses that SG FQDN as its domain name, and maps to the “UC Signaling End Point” function within the SG | 5.3.2.10.3 | Y | C | |
| 92 | The CCA shall support assignment of the following items to the EBC:<br>• Only one EBC IP address (this one IP address may be implemented in the EBC as either a single logical IP address or a single physical IP address),<br>• An EBC FQDN that maps to that IP address, and<br>• An EBC SIP URI that uses that EBC FQDN as its domain name, and maps to the “SIP B2BUA” function within the EBC. | 5.3.2.10.3 | Y | Y | |
| 93 | When directing VoIP sessions to other network appliances providing voice and video services across the DISN, the CCA shall direct these VoIP sessions to the EBC, so that the EBC can process them before directing them to the network appliances on the DISN WAN. | 5.3.2.10.4 | Y | Y | |
| 94 | When accepting VoIP sessions from other network appliances on the DISN, the CCA shall accept these VoIP sessions from the EBC, because the EBC relays them from the network appliances on the DISN WAN. | 5.3.2.10.4 | Y | Y | |
| 95 | The LSC and MFSS CCA shall meet all the requirements in Section 5.3.2.2.2.3, ASAC – Open Loop.<br>The LSC and MFSS CCA shall meet all the requirements in Section 5.3.4.10, Precedence and Preemption.<br>The LSC and MFSS CCA shall meet all the requirements in Section 5.3.4.11, Policing of Call Count Thresholds. | 5.3.2.10.5 | Y | Y | |
| 96 | The CCA shall generate a redirecting number each time it forwards a VoIP or Video session request as part of a Call Forwarding feature. | 5.3.2.10.6 | Y | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 97 | It is expected that all Assured Services products, such as LSCs and MFSSs, will support vendor-proprietary VVoIP features and capabilities, in addition to supporting the required VVoIP features and capabilities that are listed. | 5.3.2.10.6 | Y | Y | |
| 98 | The CCA shall relay received SIP and TLS authentication credentials and encryption key information from sending end systems (i.e., users, PEIs, AEIs, and EBCs) to the Information Assurance function to support the Information Assurance function's user, PEI, AEI, and EBC authentication capabilities, and its PEI, AEI, and EBC signaling stream encryption capabilities. | 5.3.2.10.7 | Y | Y | |
| 99 | The CCA MGC shall relay received H.248 and IPSec (or proprietary-protocol-equivalent) authentication credentials and encryption key information from sending end systems (i.e., MGs and SGs) to the Information Assurance function to support the Information Assurance function's MG and SG authentication capabilities, and its MG and SG signaling stream encryption capabilities. | 5.3.2.10.7 | Y | Y | |
| 100 | The CCA shall relay authentication credentials received in a SIP or AS-SIP REGISTER message from an PEI, AEI, or EBC to the Information Assurance function. | 5.3.2.10.7 | Y | Y | |
| 101 | The CCA shall relay TLS encryption key information received from a PEI or AEI to the Information Assurance function so the Information Assurance function can verify that this encryption key information can be used on the signaling streams for Voice or Video sessions to/from that PEI or AEI. | 5.3.2.10.7 | Y | Y | |
| 102 | The CCA shall relay TLS encryption key information received from an EBC to the Information Assurance function so the Information Assurance function can verify that this encryption key information can be used on the signaling streams for the Voice or Video sessions to/from that EBC. | 5.3.2.10.7 | Y | Y | |
| 103 | The CCA within the appliance shall support all Information Assurance Appliance requirements in Section 5.4, Information Assurance Requirements, which involve the appliance's SCS functions and the appliance's MGC. | 5.3.2.10.7 | Y | Y | |
| 104 | The CCA shall support supplier-proprietary Voice and Video EIs, using EI-CCA protocols that are proprietary to the LSC or MFSS supplier. | 5.3.2.10.10 | Y | Y | |
| 105 | When the CCA IWF supports AS-SIP Voice and Video AEIs, the IWF shall support these AEIs using the set of AS-SIP protocol requirements in Section 5.3.2.22, Generic AS-SIP End Instrument and Video Codec Requirements, and Section 5.3.4, AS-SIP Requirements. | 5.3.2.10.10 | Y | Y | |
| 106 | The Appliance CCA (i.e., LSC or MFSS) shall support both assured Voice and Video services. The CCA shall support both assured Voice and assured Video sessions, and shall support these sessions from both VoIP EIs and Video EIs, as described in UCR 2008, Section 5.3.2.10.10, CCA Interactions with End Instrument(s). | 5.3.2.10.11 | Y | Y | |
| 107 | The Appliance CCA shall support common procedures and protocol for VoIP and Video session control. | 5.3.2.10.11 | Y | Y | |
| 108 | The Appliance CCA shall support common procedures and protocol for feature control, for the features and capabilities given in Table 5.3.2.2-1, Assured Services Product Features and Capabilities. | 5.3.2.10.11 | Y | Y | |
| 109 | On calls to and from Proprietary VoIP and Proprietary Video EIs, the CCA shall use the appropriate parameters within the appliance supplier's Proprietary protocol messages to differentiate Proprietary VoIP sessions from Proprietary Video sessions. | 5.3.2.10.11 | Y | Y | |
| 110 | When AS-SIP EIs are supported on calls to and from AS-SIP EIs, the CCA shall use the SDP message bodies in AS-SIP INVITE, UPDATE, REFER, and ACK messages, as well as the SDP message bodies in AS-SIP 200 OK responses and earlier 1xx provisional responses, to differentiate AS-SIP Voice sessions from AS-SIP Video sessions. | 5.3.2.10.11 | Y | Y | |
| 111 | The CCA shall track VoIP sessions against corresponding Appliance VoIP budgets, and shall separately track Video sessions against corresponding Video budgets. The CCA shall maintain the Appliance's VoIP budgets separate from the Appliance's Video budget. | 5.3.2.10.11 | Y | Y | |
| 112 | As part of LSC-Level ASAC and WAN-Level ASAC Policing, the CCA shall support PBAS/ASAC for both VoIP sessions and Video sessions. | 5.3.2.10.11 | Y | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 113 | The CCA shall allow an individual EI to support both VoIP and Video sessions. The CCA shall allow an individual EI to have both VoIP and Video sessions active at the same time. | 5.3.2.10.11 | Y | Y | |
| 114 | The CCA shall support the routing of both VoIP and Video session requests from LSCs to MFSSs, from MFSSs to LSCs, and from MFSSs to MFSSs, using AS-SIP. The CCA shall direct outgoing VoIP and Video session requests to EBCs, and shall accept incoming VoIP and Video session requests from EBCs, consistent with this LSC-to-MFSS routing, MFSS-to-LSC routing, and MFSS-to-MFSS routing. | 5.3.2.10.11 | Y | Y | |
| 115 | The CCA shall support the ability to remove VoIP and Video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions. | 5.3.2.10.12 | Y | Y | |
| 116 | The MG supports interconnection of VoIP, FoIP, and MoIP media streams with the following LSC functions and end-user devices:<br>a. The LSC media server, which provides tones and announcements for LSC calls and LSC features<br>b. AS-SIP VoIP, FoIP, and MoIP AEIs on the LSC | 5.3.2.12.3.1 | | Y | |
| 117 | The MFSS MG shall be able to support MG trunk groups (referred to as internal MG connections) that either interconnect the SS (VoIP) side of the MFSS with the EO or Tandem functions on the TDM side of the MFSS. | 5.3.2.12.3.2.1 | Y | | |
| 118 | On incoming call requests to a TDM trunk group, where the CCA/MGC applies a CAC Call Denial treatment to that call request, the MG shall connect the TDM called party on the incoming call request to the appropriate CAC Call Denial tone or announcement when instructed to do so by the MGC. | 5.3.2.12.4.1.1 | Y | Y | |
| 119 | On incoming calls or call requests to a TDM trunk group, where the CCA/MGC applies an ASAC Call Preemption treatment to that call or call request, the MG shall connect the TDM calling party on the incoming call or call request to the appropriate ASAC Call Preemption tone or announcement when instructed to do so by the MGC. | 5.3.2.12.4.1.1 | Y | Y | |
| 120 | On outgoing calls or call requests from a TDM trunk group, where the CCA/MGC applies an ASAC Call Preemption treatment to that call or call request, the MG shall connect the TDM called party on the outgoing call or call request to the appropriate ASAC Call Preemption tone or announcement when instructed to do so by the MGC. | 5.3.2.12.4.1.1 | Y | Y | |
| 121 | Each MG within an appliance shall support all the appliance requirements in Section 5.4, Information Assurance Requirements, that involve an Appliance MG. | 5.3.2.12.4.2 | Y | Y | |
| 122 | When instructed to do so by the MGC, the MG shall direct TDM calls and call requests to the media server. | 5.3.2.12.4.3 | Y | Y | |
| 123 | Since each Appliance MG is an IP endpoint on the Appliance LAN, each MG shall support assignment of the following items to itself:<br>• Only one MG IP address (This one IP address may be implemented in the CCA as either a single logical IP address or a single physical IP address.)<br>• An MG FQDN that maps to that IP address<br>• An MG SIP URI that uses that MG FQDN as its domain name, and maps to a "SIP User Agent" function within the MG. | 5.3.2.12.4.4 | Y | Y | |
| 124 | The MG shall interact with the Transport Interface functions in the appliances when the MG uses the native LAN protocols, IP, and UDP to exchange SRTP media streams with PEIs, AEIs, other MGs, and the EBC over the Appliance LAN | 5.3.2.12.4.4 | Y | Y | |
| 125 | When sending VoIP media streams to PEIs or AEIs and MGs served by other network appliances, the MG shall direct these VoIP media streams to the EBC so the EBC can process them before sending them on to the remote PEIs or AEIs and MGs via the DISN WAN. | 5.3.2.12.4.5 | Y | Y | |
| 126 | When accepting VoIP media streams from PEIs or AEIs and MGs served by other network appliances, the MG shall accept these VoIP media streams from the appliance EBC, because the EBC relays them from the DISN WAN and the remote PEIs or AEIs and MGs on the DISN WAN. The MG shall recognize and act on the network-level IP addresses of the remote PEIs or AEIs and MGs, when accepting the VoIP sessions through the EBC from the DISN WAN and the remote PEIs or AEIs and MGs. | 5.3.2.12.4.5 | Y | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 127 | The MG shall support the exchange of VoIP media streams with the following voice PEIs and AEIs both on the local appliance and on remote network appliances:<br>a. Supplier-proprietary voice PEIs<br>b. Voice SIP EIs, when the appliance supplier supports these EIs<br>c. Voice H.323 EIs, when the appliance supplier supports these EIs<br>d. Voice AS-SIP AEIs | 5.3.2.12.4.8 | Y | Y | |
| 128 | The MG shall support the operation of the following features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches today:<br>• Call Hold<br>• Music on Hold<br>• Call Waiting<br>• Precedence Call Waiting<br>• Call Forwarding Variable<br>• Call Forwarding Busy Line<br>• Call Forwarding No Answer<br>• Call Transfer<br>• Three-Way Calling<br>• Hotline Service<br>• Calling Party and Called Party ID (number only)<br>• Call Pickup | 5.3.2.12.4.9 | Y | Y | |
| 129 | Each appliance MG shall support TDM trunk groups that can interconnect with the following NEs in DoD networks, in the United States and worldwide:<br>• PBXs<br>• SMEOs<br>• EOs<br>• MFSs<br>Media Gateway support for these TDM trunk groups shall be identical to the support for these trunk groups in DoD TDM PBXs, EOs, Tandem switches, and MFSs today, as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features. | 5.3.2.12.5 | Y | Y | |
| 130 | Each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the United States, including CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Media Gateway support for these TDM trunk groups to the U.S. PSTN shall be identical to the support for these trunk groups in DoD TDM PBXs, EOs, Tandem Switches, and MFSs today, as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features. | 5.3.2.12.7 | Y | Y | |
| 131 | The MG shall support foreign country ISDN PRI, where the MG handles both the media channels and the signaling channel:<br>1. For interconnection with a foreign country PSTN using foreign country ISDN PRI, from the country where the DoD user's B/P/C/S is located.<br>2. Support for ETSI PRI is required on LSC trunk groups when the LSC is used in OCONUS ETSI-compliant countries.<br>3. Support for ETSI PRI is required on MFSS trunk groups when the MFSS is used in OCONUS ETSI-compliant countries.<br>4. Support for MLPP using ISDN PRI is not required on the above trunk groups. | 5.3.2.12.8 | Y | Y | |
| 132 | The MG shall support ISDN PRI trunk groups that carry the U.S./National ISDN version of the ISDN PRI protocol. The MG shall support these U.S. PRI trunk groups conformant with the detailed U.S. ISDN PRI requirements. | 5.3.2.12.10 | Y | Y | |
| 133 | The MG shall support multiple U.S. PRI trunk groups based on the needs of the DoD user deploying the appliance. The MG shall allow each U.S. PRI trunk group at the MG to connect to: TDM EO and tandem components of the local MFSS; a different U.S. PSTN TDM NE (e.g., PBX, TDM switch); a different DoD TDM NE (e.g., PBX, TDM switch); or a different DoD IP NE (e.g., LSC, MFSS), based on the interconnection needs of the DoD user. | 5.3.2.12.10 | Y | Y | |
| 134 | The MG shall support reception of ISDN PRI messages from the CCA MGC and transmission of ISDN PRI messages to the CCA MGC. | 5.3.2.12.10 | Y | Y | |
| 135 | The MG shall connect to the ASLAN of the appliance using the physical layer and data link layer protocols of the ASLAN. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as a | 5.3.2.12.12.1 | Y | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| | physical layer and data link layer endpoint on a LAN switch in the ASLAN. | | | | |
| 136 | The MG shall connect to the ASLAN of the appliance using the IP as a Network Layer Protocol. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as an IP endpoint on an IP router on the ASLAN | 5.3.2.12.12.2 | Y | Y | |
| 137 | The MG shall support IPv4 as a Network Layer Protocol. The MG shall also support IPv6 as a Network Layer Protocol. | 5.3.2.12.12.2 | Y | Y | |
| 138 | Conformant with Section 5.3.5, IPv6 Requirements, the MG shall support dual IPv4 and IPv6 stacks (i.e., support both IPv4 and IPv6 in the same IP end point) as described in RFC 4213. | 5.3.2.12.12.2 | Y | Y | |
| 139 | The MG shall support exchange of VoIP media streams with appliance PEIs/AEIs, other appliance MGs, and the appliance EBC (and through the appliance EBC, with other PEIs/AEIs and MGs on other network appliances) using the following IETF-defined Media Transfer Protocols:<br>• SRTP, conformant with RFC 3711<br>• SRTCP, conformant with RFC 3711 | 5.3.2.12.12.4 | Y | Y | |
| 140 | The MG shall secure all VoIP media streams exchanged with appliance PEIs/AEIs, other appliance MGs, and the appliance EBC (and through the EBC, with PEIs/AEIs and MGs on other network appliances) using SRTP and SRTCP. | 5.3.2.12.12.4 | Y | Y | |
| 141 | The MG shall use UDP as the underlying Transport Layer Protocol, and IP as the underlying Network Layer Protocol, when SRTP is used for media stream exchange. | 5.3.2.12.12.4 | Y | Y | |
| 142 | When the VoIP signaling streams contain supplier-proprietary protocol messages instead of H.248 or ISDN PRI messages, the MG shall secure the proprietary protocol message exchange with the MGC using mechanisms that are as strong as, or stronger than, the use of IPSec to secure H.248 and PRI message exchange. | 5.3.2.12.12.5 | Y | Y | |
| 143 | The MG shall support TDM voice streams using the following:<br>• ITU-T 64 kbps G.711 µ-law PCM over digital trunks<br>• ITU-T 64 kbps G.711 A-law PCM over digital trunks<br>• North American 56 kbps G.711 µ-law PCM over digital trunks<br>• North American analog voice transmission over analog trunks on TDM trunk groups on the TDM side of the MG | 5.3.2.12.12.6.5 | Y | Y | |
| 144 | The MG shall convert between North American 56 kbps G.711 µ-law PCM and ITU-T 64 kbps G.711 µ-law PCM in cases where North American 56 kbps TDM voice trunks are used on the TDM side of the MG. | 5.3.2.12.12.6.5 | Y | Y | |
| 145 | The MG shall convert between North American analog voice transmission and ITU-T 64 kbps G.711 µ-law PCM in cases where North American analog voice trunks are used on the TDM side of the MG. | 5.3.2.12.12.6.5 | Y | Y | |
| 146 | The MG shall support uncompressed, packetized VoIP streams using ITU-T Recommendation G.711 µ-law PCM and ITU-T Recommendation G.711 A-law PCM (ITU-T Recommendation G.711, November 1998, plus Appendix I, September 1999, and Appendix II, September 2000) over the IP network on the VoIP side of the MG. | 5.3.2.12.12.6.5.1 | Y | Y | |
| 147 | The MG shall packetize/depacketize G.711 media streams received or sent between its TDM side and its VoIP side. | 5.3.2.12.12.6.5.1 | Y | Y | |
| 148 | The MG shall transport each packetized G.711 VoIP stream to and from the destination local PEI, local AEI, local MG, remote PEI (via an EBC), remote AEI (via an EBC), or remote MG (via an EBC) using SRTP, UDP, and IP protocol layers on the VoIP side of the MG. | 5.3.2.12.12.6.5.1 | Y | Y | |
| 149 | The MG shall support the use of uncompressed, packetized G.711 µ-law and A-law VoIP media streams for both Fixed and Deployable applications. | 5.3.2.12.12.6.5.1 | Y | Y | |
| 150 | The MG shall provide an EC capability with an echo path capacity (echo tail length) of at least 64 ms. | 5.3.2.12.13.2.2 | Y | Y | |
| 151 | The MG shall provide echo cancellation for voice, G3 Fax, and VBD modem fax calls. | 5.3.2.12.13.2.2 | Y | Y | |
| 152 | Each MG EC shall be equipped with an "echo canceller disabling signal" tone detector. This tone detector shall detect and respond to an in-band EC disabling signal from an end user's G3 Fax or VBD modem | 5.3.2.12.13.2.2 | Y | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| | device. The EC disabling signal detected shall consist of a 2100-Hz tone with periodic phase reversals inserted in that tone. | | | | |
| 153 | The MG tone detector/EC disabler shall detect the “echo canceller disabling signal” and disable the MG EC when, and only when, that signal is present for G3 Fax or VBD modem. | 5.3.2.12.13.2.2 | Y | Y | |
| 154 | The MG shall derive its clock timing from a designated T1 or PRI interface. | 5.3.2.12.14 | Y | Y | |
| 155 | The MGC within the CCA shall be responsible for controlling all the MGs within the LSC or MFSS. | 5.3.2.12.15 | Y | Y | |
| 156 | The MGC within the CCA shall be responsible for controlling all the trunks (i.e., DoD CCS7, PRI, or CAS) within each MG within the LSC or MFSS. | 5.3.2.12.15 | Y | Y | |
| 157 | The MGC within the CCA shall be responsible for controlling all media streams on each trunk within each MG. | 5.3.2.12.15 | Y | Y | |
| 158 | The MGC within the CCA shall accept IP signaling streams from an MG, conveying received PRI or CAS trunk signaling. The MGC shall return IP signaling streams to the MG accordingly, for conversion to transmitted PRI or CAS trunk signaling. | 5.3.2.12.15 | Y | Y | |
| 159 | Within the appliance (i.e., LSC or MFSS), the MGC shall use either ITU-T Recommendation H.248 (Gateway Control Protocol Version 3) or a supplier-proprietary protocol to accomplish the MG, trunk, and media stream controls described previously. | 5.3.2.12.15 | Y | Y | |
| 160 | Whenever the MG uses ITU-T Recommendation V.150.1, the following applies:<br>ITU-T Recommendation V.150.1 provides for three states: audio, VBD, and modem relay. After call setup, inband signaling may be used to transition from one state to another. In addition, V.150.1 provides for the transition to FoIP using Fax Relay per ITU-T Recommendation T.38. | 5.3.2.12.16 | Y | Y | |
| 161 | The LSC MG, MFSS MG, and WAN SS MG shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within that MG. | 5.3.2.12.17 | Y | Y | |
| 162 | The precedence level and dialed number input to the PEI or AEI shall be as specified in UCR 2008, Section 5.2.3.5.1.2, Interswitch and Intraswitch Dialing. | 5.3.2.16 | Y | Y | |
| 163 | The DSN Worldwide Numbering and Dialing Plan will be used as the addressing schema within the current DSN and its migration into the SIP environment. | 5.3.2.16.1 | Y | Y | |
| 164 | The CCA shall allow session requests from LSC, MFSS EIs, other appliances, and MFSS MGs to contain<br>• Called addresses including DSN numbers from the DSN numbering plan<br>• Called addresses including E.164 numbers from the E.164 numbering plan | 5.3.2.16.1 | Y | Y | |
| 165 | When a session request’s called address includes a DSN number from the DSN numbering plan, the CCA shall determine whether the called DSN number is local to the LSC or MFSS, or external to the LSC or MFSS. | 5.3.2.16.1 | Y | Y | |
| 166 | When a session request’s called address includes an E.164 number from the E.164 numbering plan, the CCA shall determine whether the called E.164 number is local to the LSC or MFSS, or external to the LSC or MFSS. | 5.3.2.16.1 | Y | Y | |
| 167 | The CCA shall allow each VoIP and Video PEI and AEI served by an LSC or MFSS to have both a DSN number assigned and an E.164 number assigned. | 5.3.2.16.1.1 | Y | Y | |
| 168 | For VoIP and Video PEIs or AEIs that have both a DSN number and an E.164 number assigned, the CCA shall be able to match each PEI’s or AEI’s DSN number with its E.164 number, and to match each PEI’s or AEI’s E.164 number with its DSN number. | 5.3.2.16.1.1 | Y | Y | |
| 169 | The CCA shall be able to distinguish DSN called numbers from E.164 called numbers when processing VoIP and Video session requests | 5.3.2.16.1.2 | Y | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| | from PEIs, AEIs, EBCs, MG line cards, and MG trunk groups. | | | | |
| 170 | The CCA shall be able to distinguish local [DSN or E.164] called numbers from external [DSN or E.164] called numbers when processing VoIP and Video session requests from PEIs, AEIs, EBCs, MG line cards, and MG trunk groups. | 5.3.2.16.1.2 | Y | Y | |
| 171 | The MFSS or LSC is only required to support one network FQDN for use with SIP URI domain names: “uc.mil” if that appliance is used for SBU traffic, and “cuc.mil” if that appliance is used for classified traffic. | 5.3.2.16.1.4.1 | Y | Y | |
| 172 | The MFSS or LSC is required to ensure that all AS-SIP session requests entering or leaving that appliance use the network FQDN of that appliance (i.e., “uc.mil” for SBU traffic, or “cuc.mil” for Classified traffic) as the domain name in called SIP URIs. | 5.3.2.16.1.4.1 | Y | Y | |
| 173 | All voice systems, TDM or IP technology-based, must contain subscriber assignment information. | 5.3.2.16.1.5 | Y | Y | |
| 174 | Use of the Commercial Cost Avoidance functionality shall be an optional application that can be configured (i.e., enabled and disabled) on each RTS LSC. | 5.3.2.23 | | Y | |
| 175 | The LSC shall be able to query the DISN RTS Routing Database on “99 dialed PSTN number” call requests from LSC end users. When the database responds to this query with a DSN number that matches the dialed PSTN number, the LSC shall route the call request over the appropriate IP (AS-SIP) or TDM (e.g., T1.619A PRI) path, using the DSN number returned by the database. When the database responds with a “number not found” indication, the LSC shall route the call request to the local TDM PSTN trunk group (PRI or CAS) on the LSC’s MG, using the originally dialed commercial number. | 5.3.2.23 | | Y | |
| 176 | The query-response interface between the LSC and the RTS Routing Database shall be LDAP Version 3 (v3) over TLS over IP. This LDAPv3 interface shall be compliant with RFC 4510. | 5.3.2.23 | | Y | |
| 177 | The encoding of the LDAPv3 messages and data schema used on the DB query interface between the LSC and the RTS Routing Database shall follow the BER of ASN.1, consistent with Section 5.1, Protocol Encoding, of RFC 4511. | 5.3.2.23 | | Y | |
| 178 | The DB query interface between the LSC and the RTS Routing Database shall traverse the data firewalls (and not the RTS EBC firewalls) at both the LSC and RTS Routing Database sites. | 5.3.2.23 | | Y | |
| 179 | After transmitting a Commercial Cost Avoidance query to the Database, the LSC shall start a “Commercial Cost Avoidance Query Response” timer awaiting a Database response. If the timer expires and no response is received, the LSC shall route the call request to the local TDM PSTN trunk group (PRI or CAS) using the originally dialed commercial number. | 5.3.2.23 | | Y | |
| 180 | On Commercial Cost Avoidance call requests that are re-routed to DSN numbers by the database, the LSC shall respond to MFSS or WAN SS AS-SIP signaling indicating that the call was rejected (i.e., an AS-SIP 4xx, 5xx, or 6xx response to an AS-SIP INVITE message), by overflowing these calls from the AS-SIP trunk group to the local TDM PSTN trunk group (PRI or CAS) using the originally dialed commercial number. | 5.3.2.23 | | Y | |
| 181 | On Commercial Cost Avoidance call requests that are re-routed to DSN numbers by the database, the MFSS or WAN SS shall accept AS-SIP call requests from the LSC where the DSN number is identified as the called number. The MFSS or WAN SS shall also be capable of returning AS-SIP signaling to the calling LSC that indicates “404 Not Found,” “480 Temporarily Unavailable,” or “500 Server Internal Error.” The MFSS or WAN SS shall be capable of generating this AS-SIP signaling on its own, and shall be capable of relaying that AS-SIP signaling when it is received from a remote MFSS, remote WAN SS, or remote LSC. | 5.3.2.23 | Y | | Y |
| 182 | For each RTS end user served by an LSC, the LSC shall be able to upload that user’s DSN phone number, PSTN phone number, and a unique LSC CCA-ID, Primary MFSS/WAN SS CCA-ID, and Backup MFSS/WAN SS CCA-ID to the RTS Routing Database. | 5.3.2.23 | | Y | |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 183 | The AS-SIP signaling appliance shall divert ALL unanswered RTS VoIP calls above the ROUTINE precedence level to a designated RTS DN for PCD (e.g., the number of an attendant console or group of attendant consoles). | 5.3.2.25 | Y | Y | C |
| 184 | Unanswered RTS VoIP calls above the ROUTINE precedence level shall not be forwarded to voicemail, and shall not be forwarded to ACD systems. Instead, they should divert to the PCD DN when the PCD time period expires. | 5.3.2.25 | Y | Y | C |
| 185 | Unanswered RTS VoIP calls at the ROUTINE precedence level shall still be forwarded to voicemail or to ACD systems (when Call Forwarding Don't Answer is assigned to the called RTS DN), even though PCD is enabled and configured for the AS-SIP signaling appliance. | 5.3.2.25 | Y | Y | C |
| 186 | Calls above the ROUTINE precedence level that are destined to (directly dialed to) DNs assigned to voicemail or ACD systems shall only divert to the PCD DN as specified above (i.e., when they are unanswered at the voicemail or ACD system, and the PCD time period expires). | 5.3.2.25 | Y | Y | C |
| 187 | ROUTINE precedence level calls that are destined to (directly dialed to) DNs assigned to voicemail or ACD systems shall be allowed. | 5.3.2.25 | Y | Y | C |
| 188 | Incoming precedence calls to the attendant's listed DN, and incoming calls that are diverted to this attendant DN, shall be placed in a queue for the attendant console (or group of attendant consoles). | 5.3.2.25 | Y | Y | C |
| 189 | When a group of attendant consoles on the same LSC is used, and calls are either placed or diverted to the attendant console DN, call distribution across the Console Group shall be used to reduce excessive caller waiting times. | 5.3.2.25 | Y | Y | C |
| 190 | Incoming calls (placed and diverted) to the console DN shall be queued for attendant service by call precedence and time of arrival. The highest precedence call with the longest holding time in the queue shall be offered to an attendant first. | 5.3.2.25 | Y | Y | C |
| 191 | A recorded message of explanation (e.g., ATQA) shall be applied automatically to all the waiting calls in the Attendant Console queue (refer to Table 5.3.4-9, Announcements). | 5.3.2.25 | Y | Y | C |
| 192 | The RTS Attendant Console shall interoperate with PBAS/ASAC as described in<br>• Section 5.3.2.7.2.1, PBAS/ASAC Requirements<br>• Section 5.3.2.2.2.3, ASAC – Open Loop<br>• Section 5.3.4.10, Precedence and Preemption<br>The console shall be able to initiate all levels of RTS precedence calls (i.e., ROUTINE through FLASH-OVERRIDE). | 5.3.2.26.1 | Y | Y | C |
| 193 | The RTS Attendant Console shall provide a visual display of each precedence level and the calling number, for incoming direct dialed calls to the attendant, and diverted calls to the attendant (e.g., calls that reach the attendant through PCD). | 5.3.2.26.2 | Y | Y | C |
| 194 | The appliance and the attendant console shall give the attendant the ability to verify and override a busy line condition. | 5.3.2.26.4 | Y | Y | C |
| 195 | If the attendant uses BLV on a called line, and that called line (called EI) is busy, the appliance and the attendant console shall give an audible and visual "called line busy" indication back to the attendant. | 5.3.2.26.4 | Y | Y | C |
| 196 | The appliance and the attendant console shall prevent an attendant from activating BLV or Emergency Interrupt to called lines and called numbers that are located in the commercial network (the PSTN). | 5.3.2.26.4 | Y | Y | C |
| 197 | The appliance and the attendant console shall give the attendant the ability to use Emergency Interrupt to interrupt an existing call on a busy line, and inform the busy user of a new incoming call. | 5.3.2.26.4 | Y | Y | C |
| 198 | The appliance shall give selected destination EIs the ability to be exempt from Emergency Interrupt and attendant break-in. | 5.3.2.26.4 | Y | Y | C |
| 199 | The appliance and the attendant console shall have the ability to route all calls that are normally directed to the console to a separate night service deflection number. The night service deflection number shall be a fixed (preconfigured) or manually-selected DN. | 5.3.2.26.5 | Y | Y | C |
| 200 | When an attendant redirects an incoming call to a destination station, and that station is either busy or does not answer the call within a preset time, the appliance and the attendant console shall ensure that | 5.3.2.26.6 | Y | Y | C |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| | calling party on the redirected call is recalled automatically to the console. In this case, the appliance shall ensure that that the "recalled" call is returned to the console that originally processed the call. | | | | |
| 201 | The appliance and the attendant console shall have the ability to place calls (both directed to the attendant and diverted to the attendant) into a waiting queue. | 5.3.2.26.7 | Y | Y | C |
| 202 | The appliance and the attendant console shall ensure that calls in the attendant queue are not lost when a console is placed out of service or has its calls forwarded to a night service deflection number. | 5.3.2.26.7 | Y | Y | C |
| 203 | The LSCs that support SIP EIs MUST comply with the differentiated set of requirements defined for SIP EIs if they serve SIP EIs, and LSCs MUST comply with the differentiated set of requirements defined for H.323 and/or vendor-proprietary EIs if they serve H.323 and/or vendor-proprietary EIs. | 5.3.4.7.1 | | Y | |
| 204 | All AS-SIP signaling appliances MUST comply with the SIP syntax and encoding rules set forth in RFC 3261. [RFC 3261, Section 25, Augmented BNF for the SIP Protocol] | 5.3.4.7.1.1 | Y | Y | Y |
| 205 | When an AS-SIP signaling appliance does not understand a header field in a request (and support for the header field is not a mandatory requirement under this specification), the AS-SIP signaling appliance MUST ignore that header field and continue processing the message. The AS-SIP signaling appliances MUST ignore any malformed header fields that are not necessary for processing requests. | 5.3.4.7.1.3 | Y | Y | Y |
| 206 | When an AS-SIP signaling appliance, that is implemented as a SIP proxy, receives a SIP Request message, 2xx response, or 18x response, then the AS-SIP signaling appliance MUST add a Record-Route header whereby the userinfo part of the SIP URI is a unique identifier for the AS-SIP signaling appliance and an IP address is used for the host name. | 5.3.4.7.1.3c | Y | Y | Y |
| 207 | All AS-SIP signaling appliances MUST be call stateful. | 5.3.4.7.1.4 | Y | Y | Y |
| 208 | Upon receipt of a new request, AS-SIP signaling appliances MUST perform request validation, route information preprocessing, determine request targets, perform request forwarding, perform response processing, process timer C, handle transport error, handle CANCEL processing, and perform proxy route processing according to RFC 3261 | 5.3.4.7.1.5 | Y | Y | Y |
| 209 | All AS-SIP signaling appliances MUST support generation of the long form of the SIP header fields along with the receipt and processing of the long form of the SIP header fields. | 5.3.4.7.1.7 | Y | Y | Y |
| 210 | All AS-SIP signaling appliances MUST support receiving and processing the compact form of the SIP header fields. | 5.3.4.7.1.8 | Y | Y | Y |
| 211 | All AS-SIP signaling appliances serving IP EIs MUST support the offer/answer model for the Session Description Protocol (SDP). | 5.3.4.7.1.9 | Y | Y | Y |
| 212 | If an LSC receives a call request from a served IP EI and the LSC has been unable to establish a TLS connection with its EBC and is unable to do so upon receipt of the INVITE, then the AS-SIP signaling appliance MUST ensure that the IP EI plays the Isolated Code Announcement (ICA) and terminates the call request and MUST send an alarm to the NMS. | 5.3.4.7.1.10 | | Y | |
| 213 | When an SS receives an INVITE from either a served LSC or another SS where the Request-URI has a DSN telephone number for which the SS has no entry in its Location Server, then the SS MUST respond with a 404 (Not Found) response code. | 5.3.4.7.1.12 | Y | | Y |
| 214 | When an LSC receives an inbound INVITE from its primary (or secondary) SS whose Request-URI has a DSN telephone number for which the LSC has no entry in its Location Server, then the LSC MUST respond with a 404 (Not Found) response message. | 5.3.4.7.1.13 | | Y | |
| 215 | The LSCs serving IP EIs MUST ensure that all outbound INVITEs forwarded onto the UC WAN include a Supported header with the option tag "100rel." | 5.3.4.7.1.14 | | Y | |
| 216 | When an AS-SIP signaling appliance receives an INVITE (having an sdp offer) and will be interworking the signaling to the TDM network, then the AS-SIP signaling appliance MUST return an sdp answer in the first non-failure reliable provisional response. | 5.3.4.7.1.15 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 217 | When an LSC receives an INVITE (having an sdp offer) intended for a served IP EI, then the AS-SIP signaling appliance MUST NOT return an sdp answer in any provisional response and MUST only place the sdp answer in the 200 response. | 5.3.4.7.1.16 | | Y | |
| 218 | When an AS-SIP signaling appliance receives an Empty INVITE (i.e., an INVITE that does not include an sdp offer) and said AS-SIP signaling appliance will be interworking the signaling to the TDM network, then the AS-SIP signaling appliance MUST send an sdp offer in the first reliable non-failure provisional response (1xx response code greater than a 100 response code). | 5.3.4.7.1.17 | Y | Y | Y |
| 219 | When an AS-SIP signaling appliance receives an Empty INVITE intended for a served IP EI, then the AS-SIP signaling appliance MUST NOT send an sdp offer in any provisional response (1xx response code greater than a 100 response code) and MUST only send the sdp offer in the 200 response. | 5.3.4.7.1.18 | Y | Y | Y |
| 220 | When an AS-SIP signaling appliance that is interworking SIP signaling with the TDM network receives a 180 (Ringing) response from the IP network, the AS-SIP signaling appliance MUST ensure that the appropriate ring back tone (e.g., ring back, precedence ring back) is generated on the TDM network. | 5.3.4.7.1.19 | Y | Y | Y |
| 221 | Announcements are not sent in-band on the DSN TDM network; therefore, when an AS-SIP signaling appliance that is interworking SIP signaling with the TDM network receives a 480 (Temporarily Unavailable), 486 (Busy Here), or 488 (Not Acceptable Here) response from the IP network with either no Reason header or a Reason header that does NOT have a preemption cause, the AS-SIP signaling appliance does NOT generate an announcement to be sent to the TDM network, rather it sends either a REL with Q.850 cause code 46 precedence call blocked (in the case of SS7 ISUP) or a Disconnect (in the case of ISDN) with the appropriate cause code message to the TDM network. | 5.3.4.7.1.20 | Y | Y | Y |
| 222 | An LSC that receives an outbound call request from a served IP EI MAY include an audio media feature tag and a video media feature tag, as appropriate, in the Contact header field of the INVITE message. | 5.3.4.7.1.21 | | Y | |
| 223 | The AS-SIP signaling appliances are NOT required to process and act on the audio media tag and the video media tag in the Contact header but all intermediary AS-SIP signaling appliances MUST preserve the audio media tag (if present) and the video tag (if present) when forwarding the INVITE. (i.e., intermediary AS-SIP signaling appliances MUST NOT strip off or modify the media feature tags). | 5.3.4.7.1.22 | Y | Y | Y |
| 224 | When an LSC receives a call request from a served IP EI intended for a destination outside the enclave, then the AS-SIP signaling appliance MUST generate the P-Asserted-Identity header. | 5.3.4.7.1.23 | | Y | |
| 225 | The LSC serving the AS-SIP EI MUST support authentication of the AS-SIP EIs. The user of the AS-SIP EI is required to perform user authentication to the LSC when initiating precedence call requests. | 5.3.4.7.1.26 | | Y | |
| 226 | The LSCs serving IP EIs (other than AS-SIP EIs) MUST support authentication of the IP EIs. The user of the IP EI is required to perform user authentication to the LSC when initiating precedence call requests. | 5.3.4.7.1.31 | | Y | |
| 227 | When an LSC serving H.323 and/or proprietary EIs receives a request that contains a Require header field with one or more option tags that it does not understand, then it MUST return a 420 (Bad Extension) response code. The response MUST include an Unsupported header field listing those option tags the element did not understand. | 5.3.4.7.1.35 | | Y | |
| 228 | The LSCs and AS-SIP EIs MUST support the generating, receiving, and processing of SIP CANCEL requests. | 5.3.4.7.2.2 | | Y | |
| 229 | The AS-SIP signaling appliances MUST support the keep-alive mechanism for SIP sessions. | 5.3.4.8.1.1 | Y | Y | Y |
| 230 | The AS-SIP signaling appliances MUST support the generating, receiving, and processing of the Session-Expires and Min-SE header fields. | 5.3.4.8.1.3 | Y | Y | Y |
| 231 | The AS-SIP signaling appliances MUST support the 422 (Session Interval Too Small) response code. | 5.3.4.8.1.4 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 232 | The AS-SIP signaling appliances MUST support the option tag "timer" for use with the Supported and Require header fields; however, an AS-SIP signaling appliance acting as a UAC or a SIP EI acting as a UAC MUST NOT place the option tag "timer" in either a Require header or a Proxy-Require header. | 5.3.4.8.1.4 | Y | Y | Y |
| 233 | When an AS-SIP signaling appliance receives an outbound request from a served H.323 or proprietary IP EI, then the AS-SIP signaling appliance MUST operate in accordance with the UAC behavior (when responsible for performing the refresh). | 5.3.4.8.1.8 | Y | Y | Y |
| 234 | When an AS-SIP signaling appliance receives a call request from another AS-SIP signaling appliance, and the destination is a served H.323 or proprietary IP EI, then the AS-SIP signaling appliance MUST operate in accordance with the UAS behavior (when responsible for performing the refresh). | 5.3.4.8.1.10 | Y | Y | Y |
| 235 | When SDP information is present in a SIP message, the SIP message MUST have a content-type header having the MIME Content-Type "application/sdp". | 5.3.4.9.1.2 | Y | Y | Y |
| 236 | The SDP parser in the AS-SIP signaling appliance (and all AS-SIP EIs, including AS-SIP video conferencing EIs) MUST be able to accept and handle without error any of the SDP line types enumerated in RFC 2327 even if the application ignores the contents. | 5.3.4.9.1.3 | Y | Y | Y |
| 237 | The precedence level of the call request MUST be set forth in a SIP Resource-Priority header field whose syntax is in accordance with RFC 4412, as modified in UCR 2008, Section 5.3.4.10.2 | 5.3.4.10.2.1 | Y | Y | C |
| 238 | Video telephony EIs MUST, as the default configuration, require an end user wishing to place a call that includes video, to affirmatively signal the intention to include video to the EI every time the caller wishes to engage in a video telephony call. | 5.3.4.12.1.1 | Y | Y | C |
| 239 | Every time a caller requests a video telephony call and the video portion of the telephony call is successfully established, then the video telephony EI MUST provide the user with an affirmative confirmation that the video is enabled either before, or upon successful completion of, session establishment. | 5.3.4.12.1.2 | Y | Y | C |
| 240 | When an INVITE with an sdp offer that includes both audio and video capabilities is received by an LSC serving a destination EI that supports video telephony, then when the call request is received by the destination EI the destination EI MUST indicate to the callee that a telephony call requesting video connectivity has been received. | 5.3.4.12.2.1 | Y | Y | C |
| 241 | Every time a callee accepts a video telephony call and the video portion of the telephony call is successfully established, then the video telephony EI MUST provide the user with an affirmative confirmation that the video is enabled either before or upon successful session establishment. | 5.3.4.12.2.3 | Y | Y | C |
| 242 | AS-SIP Signaling appliances must follow call flows depicted in section 5.3.4.13 for all call features and calling services. | 5.3.4.13 | Y | Y | Y |
| 243 | AS-SIP Signaling appliances must follow requirements depicted in section 5.3.4.14 for all IP to TDM and TDM to IP translations. | 5.3.4.14 | Y | Y | Y |
| 244 | When an interworking AS-SIP signaling appliance receives a request that contains a Require header field with one or more option-tags that it does not understand, then the interworking AS-SIP signaling appliance MUST return a 420 (Bad Extension) response. The response MUST include an Unsupported header field listing those option-tags the element did not understand. | 5.3.4.16.1.1 | Y | Y | Y |
| 245 | All outbound INVITEs generated by an interworking AS-SIP signaling appliance MUST include a Supported header with the option tag "100rel." | 5.3.4.16.1.2 | Y | Y | Y |
| 246 | Interworking AS-SIP signaling appliances MUST support generating and receiving SIP INVITE requests. Interworking AS-SIP signaling appliances MUST support generating and receiving SIP re-INVITES. | 5.3.4.16.2.1 | Y | Y | Y |
| 247 | Interworking AS-SIP signaling appliances MUST support generating and receiving SIP CANCEL requests. | 5.3.4.16.2.2 | Y | Y | Y |
| 248 | Interworking AS-SIP signaling appliances MUST support generating and receiving SIP OPTIONS requests. | 5.3.4.16.2.4 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 249 | Interworking AS-SIP signaling appliances MUST support generating and receiving SIP BYE requests | 5.3.4.16.2.5 | Y | Y | Y |
| 250 | Interworking AS-SIP signaling appliances MUST support generating and receiving SIP ACK requests | 5.3.4.16.2.6 | Y | Y | Y |
| 251 | Interworking AS-SIP signaling appliances MUST support generating and receiving the SIP PRACK method. Interworking AS-SIP signaling appliances MUST support use of the option tag "100rel" with the Require header and Supported header, and MUST support the use of header fields RAck and RSeq. | 5.3.4.16.2.8 | Y | Y | Y |
| 252 | Interworking AS-SIP signaling appliances MUST support generating and receiving the SIP UPDATE method | 5.3.4.16.2.9 | Y | Y | Y |
| 253 | Inter-working AS-SIP signaling appliances MUST be capable of receiving/processing REFER requests, the Refer-To header, and the REFER event package. | 5.3.4.16.2.10 | Y | Y | Y |
| 254 | Interworking AS-SIP signaling appliances MUST support the NOTIFY method for event notification. | 5.3.4.16.2.12 | Y | Y | Y |
| 255 | Interworking AS-SIP signaling appliances MUST, in adherence with the enumerated RFCs, be capable of generating, receiving, and processing the SIP headers listed in UCR 2008 Section 5.3.4.16.3.1 | 5.3.4.16.3.1 | Y | Y | Y |
| 256 | The From header MUST include a tag field as specified in RFC 3261, Section 19.3. | 5.3.4.16.3.3 | Y | Y | Y |
| 257 | The To header of a request that is part of a dialog MUST include a tag field as specified in RFC 3261, Section 19.3. | 5.3.4.16.3.4 | Y | Y | Y |
| 258 | Interworking AS-SIP signaling appliances MUST support the use of option tags for the Require, Supported, and Unsupported headers. | 5.3.4.16.3.5 | Y | Y | Y |
| 259 | When the interworking LSC sends an initial AS-SIP INVITE to its local EBC intended for its SS, the interworking LSC MUST add two Route header field values, which either takes the form of a route set comprising two Route headers where the first Route header is the sip uri for the EBC at the enclave and the second Route header is the sip uri for the EBC serving the SS, or takes the form of one Route header with two comma-separated field values. | 5.3.4.16.3.6 | | Y | |
| 260 | When an interworking SS forwards an initial AS-SIP INVITE to a peer SS, then the interworking SS MUST add a route set comprising two Route headers where the first Route header is the SIP URI for the EBC that serves the interworking SS, and the second Route header is the SIP URI for the EBC serving the peer SS. | 5.3.4.16.3.7 | Y | | Y |
| 261 | When an interworking AS-SIP signaling appliance generates an outbound AS-SIP request, the interworking AS-SIP signaling appliance MUST add its own VIA header to the AS-SIP request. | 5.3.4.16.3.8 | Y | Y | Y |
| 262 | When an interworking AS-SIP signaling appliance receives a SIP response to be translated into TDM signaling, then the interworking AS-SIP signaling appliance operates as the UAC for SIP purposes. | 5.3.4.16.3.9 | Y | Y | Y |
| 263 | When an interworking AS-SIP signaling appliance receives an inbound SIP request to be translated into TDM signaling, then the AS-SIP signaling appliance operates as the UAS for SIP purposes. | 5.3.4.16.3.10 | Y | Y | Y |
| 264 | When an interworking AS-SIP signaling appliance generates a SIP response on behalf of a signaling message received from the TDM network, then before forwarding the SIP response the interworking AS-SIP signaling appliance MUST include the VIA headers received in the corresponding SIP request. | 5.3.4.16.3.11 | Y | Y | Y |
| 265 | When an interworking AS-SIP signaling appliance operating as an originating gateway receives an IAM from the TDM network and sends an INVITE to another AS-SIP signaling appliance (SS or LSC), then the interworking AS-SIP signaling appliance MUST add a CCA-ID parameter to the SIP URI of the Contact header populated with its unique identifier before forwarding the INVITE onward to the next AS-SIP signaling appliance. | 5.3.4.16.3.12 | Y | Y | Y |
| 266 | Interworking AS-SIP signaling appliances MUST support generating, receiving, and processing the provisional (1xx) response codes: 100 (Trying), 180 (Ringing), and 183 (Session Progress). | 5.3.4.16.4.1 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 267 | Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the successful (2xx) response codes: 200 (OK) [RFC 3261, Section 21.2, 200 OK] and 202 (Accepted) | 5.3.4.16.4.2 | Y | Y | Y |
| 268 | Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the request failure (4xx) response codes: 400 (Bad Request), 401 (Unauthorized), 403 (Forbidden), 404 (Not Found), 405 (Method Not Allowed), 406 (Not Acceptable), 407 (Proxy Authentication Required), 408 (Request Timeout), 410 (Gone), 413 (Request Entity Too Large), 414 (Request-URI Too Long), 415 (Unsupported Media Type), 416 (Unsupported URI Scheme), 417 (Unknown Resource-Priority), 420 (Bad Extension), 421 (Extension Required), 422 (Session Interval Too Small), 423 (Interval Too Brief), 480 (Temporarily Unavailable), 481 (Call/Transaction Does Not Exist), 482 (Loop Detected), 483 (Too Many Hops), 484 (Address Incomplete), 485 (Ambiguous), 486 (Busy Here), 487 (Request Terminated), 488 (Not Acceptable Here), and 491 (Request Pending). | 5.3.4.16.4.4 | Y | Y | Y |
| 269 | Interworking AS-SIP signaling appliances upon properly receiving a CANCEL request for an INVITE MUST first send a 200 (OK) response code to the CANCEL, and then follow up with a 487 (Request Terminated) response code to the INVITE. | 5.3.4.16.4.5 | Y | Y | Y |
| 270 | Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the server failure (5xx) response codes: 500 (Server Internal Error), 501 (Not Implemented), 502 (Bad Gateway), 503 (Service Unavailable), 504 (Server Timeout), 505 (Version Not Supported), 513 (Message Too Large) [RFC 3261, Section 21.5, Server Failure 5xx], and 580 (Precondition Failure) | 5.3.4.16.4.6 | Y | Y | Y |
| 271 | Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the global failures (6xx) response codes: 600 (Busy Everywhere), 603 (Decline), 604 (Does Not Exist Anywhere), and 606 (Not Acceptable). | 5.3.4.16.4.7 | Y | Y | Y |
| 272 | When an interworking AS-SIP signaling appliance receives an outbound request from the PSTN (i.e., the interworking AS-SIP signaling appliance is operating as an originating gateway) and the destination is NOT an IP EI directly served by the interworking AS-SIP signaling appliance, then the interworking AS-SIP signaling appliance MUST operate in accordance with the UAC behavior set forth in RFC 4028. | 5.3.4.17.1.1 | Y | Y | Y |
| 273 | When an interworking AS-SIP signaling appliance acting as a terminating gateway receives a call request from another AS-SIP signaling appliance, then the interworking AS-SIP signaling appliance MUST operate in accordance with the UAS behavior set forth in RFC 4028. | 5.3.4.17.1.3 | Y | Y | Y |
| 274 | Interworking AS-SIP signaling appliances MUST support the option tag “resource-priority” for use with the Require header. | 5.3.4.18.3.1 | Y | Y | Y |
| 275 | The interworking AS-SIP signaling appliance MUST receive and accept a Require header field with the option tag “resource-priority” in the INVITE, UPDATE, and REFER messages. Interworking AS-SIP signaling appliances MUST NOT reject the message with a 420 (Bad Extension) response code, but rather it MUST accept the request and translate it into the appropriate TDM signaling message as required. | 5.3.4.18.3.2 | Y | Y | Y |
| 276 | If an interworking AS-SIP signaling appliance receives an inbound ROUTINE call request over the IP network for a destination on the TDM network and the interworking AS-SIP signaling appliance has insufficient bandwidth-related resources (e.g., lack of circuit-switched trunk capacity for bearer traffic) to handle the call request, the interworking AS-SIP signaling appliance MUST reply with a 488 (Not Acceptable Here) response code and SHOULD include a Warning header with warning code 370 (Insufficient Bandwidth). | 5.3.4.18.4.5 | Y | Y | Y |
| 277 | If an interworking AS-SIP signaling appliance receives an inbound precedence call request (i.e., with precedence level PRIORITY or above) over the IP network for a destination on the TDM network and the interworking AS-SIP signaling appliance has insufficient bandwidth-related resources (e.g., lack of circuit-switched trunk capacity for bearer traffic) to handle the call request, and if there are insufficient existing calls (and/or call requests) of lower precedence whose removal would provide the necessary resources to support the pending call request, then: | 5.3.4.18.4.6 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| | - The interworking AS-SIP signaling appliance MUST reply with a 488 (Not Acceptable Here) response code and SHOULD include a Warning header with warning code 370 (Insufficient Bandwidth), and<br>- The AS-SIP signaling appliance serving the calling IP EI MUST arrange for a BPA to be played to the calling IP EI before terminating the call. | | | | |
| 278 | When an interworking AS-SIP signaling appliance receives a precedence call request from the IP network that it translates and forwards onto the TDM network and the response from the TDM network is a REL with Q.850 cause code 46 precedence call blocked (in the case of SS7 ISUP) or a Disconnect with Q.850 cause code 46 precedence call blocked (in the case of ISDN), then the interworking AS-SIP signaling appliance MUST generate a 488 (Not Acceptable Here) response that SHOULD include a "Warning" header with warning code 370 (Insufficient Bandwidth) with no Reason header that it sends onto the IP network. | 5.3.4.18.6.2 | Y | Y | Y |
| 279 | Interworking AS-SIP signaling appliances directly serving IP EIs MUST comply with the requirements in UCR 2008, Section 5.3.4.10.3.3.2, Implementing the Network Preemption. In addition, interworking AS-SIP signaling appliances directly serving IP EIs MUST meet the enumerated requirements in section 5.3.4.18.6.3.2. | 5.3.4.18.6.3 | Y | Y | Y |
| 280 | AS-SIP signaling appliances must follow all call flows depicted in UCR 2008 Section 5.3.4.19 for all supplementary services. | 5.3.4.19 | Y | Y | Y |
| 281 | The product shall support dual IPv4 and IPv6 stacks as described in RFC 4213. | 5.3.5.4 | Y | Y | Y |
| 282 | Dual stack end points or Call Control Agents shall be configured to choose IPv4 over IPv6. | 5.3.5.4 | Y | Y | Y |
| 283 | All nodes that are "IPv6-capable" shall be carefully configured and verified that the IPv6 stack is disabled until it is deliberately enabled as part of a risk management strategy. | 5.3.5.4 | Y | Y | Y |
| 284 | The product shall support the IPv6 format as described in RFC 2460 and updated by RFC 5095. | 5.3.5.4 | Y | Y | Y |
| 285 | The product shall support the transmission of IPv6 packets over Ethernet networks using the frame format defined in RFC 2464. | 5.3.5.4 | Y | Y | Y |
| 286 | The product shall support a minimum MTU of 1280 bytes. | 5.3.5.4.1 | Y | Y | Y |
| 287 | The product shall not use the Flow Label field as described in RFC 2460. The product shall be capable of setting the Flow Label field to zero when originating a packet. The product shall not modify the Flow Label field when forwarding packets. The product shall be capable of ignoring the Flow Label field when receiving packets. | 5.3.5.4.2 | Y | Y | Y |
| 288 | The product shall support the IPv6 Addressing Architecture as described in RFC 4291. | 5.3.5.4.3 | Y | Y | Y |
| 289 | The product shall support the IPv6 Scoped Address Architecture as described in RFC 4007. | 5.3.5.4.3 | Y | Y | Y |
| 290 | The product shall support Neighbor Discovery for IPv6 as described in RFC 2461 and RFC 4861. | 5.3.5.4.5 | Y | Y | Y |
| 291 | The product shall not set the override flag bit in the Neighbor Advertisement message for solicited advertisements for anycast addresses or solicited proxy advertisements. | 5.3.5.4.5 | Y | Y | Y |
| 292 | When a valid "Neighbor Advertisement" message is received by the product and the product neighbor cache does not contain the target's entry, the advertisement shall be silently discarded. | 5.3.5.4.5 | Y | Y | Y |
| 293 | When a valid "Neighbor Advertisement" message is received by the product and the product neighbor cache entry is in the INCOMPLETE state when the advertisement is received and the link layer has addresses and no target link-layer option is included, the product shall silently discard the received advertisement. | 5.3.5.4.5 | Y | Y | Y |
| 294 | When address resolution fails on a neighboring address, the entry shall be deleted from the product's neighbor cache. | 5.3.5.4.5 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 295 | The product shall support the ability to configure the product to ignore Redirect messages. The product shall only accept Redirect messages from the same router as is currently being used for that destination. | 5.3.5.4.5.1 | Y | Y | Y |
| 296 | If the product supports routing functions, the product shall inspect valid router advertisements sent by other routers and verify that the routers are advertising consistent information on a link and shall log any inconsistent router advertisements. The product shall prefer routers that are reachable over routers whose reachability is suspect or unknown. | 5.3.5.4.5.2 | Y | Y | Y |
| 297 | The product shall support manual assignment of IPv6 addresses. | 5.3.5.4.6 | Y | Y | Y |
| 298 | The product shall support the ICMPv6 as described in RFC 4443. The product shall have a configurable rate limiting parameter for rate limiting the forwarding of ICMP messages. | 5.3.5.4.7 | Y | Y | Y |
| 299 | The product shall support the capability to enable or disable the ability of the product to generate a Destination Unreachable message in response to a packet that cannot be delivered to its destination for reasons other than congestion. | 5.3.5.4.7 | Y | Y | Y |
| 300 | The product shall support the enabling or disabling of the ability to send an Echo Reply message in response to an Echo Request message sent to an IPv6 multicast or anycast address. | 5.3.5.4.7 | Y | Y | Y |
| 301 | The product shall validate ICMPv6 messages, using the information contained in the payload, before acting on them. | 5.3.5.4.7 | Y | Y | Y |
| 302 | The product shall support MLD as described in RFC 2710. | 5.3.5.4.8 | Y | Y | Y |
| 303 | For traffic engineering purposes, the bandwidth required per voice subscriber is calculated to be 110.0 kbps (each direction) for each IPv6 call. | 5.3.5.4.11 | Y | Y | Y |
| 304 | The product shall forward packets using the same IP Version as the Version in the received packet. | 5.3.5.4.12 | Y | Y | Y |
| 305 | The product shall use the Alternative Network Address Types (ANAT) semantics for the Session Description Protocol (SDP) in accordance with RFC 4091 when establishing media streams from dual-stacked appliances for AS-SIP signaled sessions. | 5.3.5.4.12 | Y | Y | Y |
| 306 | The product shall prefer any IPv4 address to any IPv6 address when using ANAT semantics. | 5.3.5.4.12 | Y | Y | Y |
| 307 | The product shall place the option tag “SDP-ANAT” in a Required header field when using ANAT semantics in accordance with RFC 4092. | 5.3.5.4.12 | Y | Y | Y |
| 308 | The products shall support Differentiated Services as described in RFC 2474 for a voice and video stream in accordance with Section 5.3.2, Assured Services Requirements, and Section 5.3.3, Network Infrastructure E2E Performance Requirements, plain text DSCP plan. | 5.3.5.4.14 | Y | Y | Y |
| 309 | The LSC must meet all requirements for FCAPS Management and audit logs as listed in UCR 2008 section 5.3.2.7.2.6 | 5.3.2.7.2.6 | | Y | |
| 310 | The physical interface between the DISA VVoIP EMS and the network components (i.e., LSC, MFSS, EBC, CE Router) is a 10/100-Mbps Ethernet interface. The interface will work in either of the two following modes using auto-negotiation: IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995. | 5.3.2.4.4 | Y | Y | Y |
| 311 | Redundant physical Ethernet interfaces are required for signaling and bearer traffic. If the primary signaling and bearer Ethernet interface fails, then traffic shall be switched to the backup signaling and bearer Ethernet interface. | 5.3.2.4.4 | Y | Y | Y |
| 312 | The MFSS shall provide a single, common interface to the DISA NMS. The single interface shall provide access to MFSS features and functions for both the TDM and SS side of the MFSS. | 5.3.2.8.3.1 | Y | | |
| 313 | The MFSS-to-NMS interface shall be an Ethernet connection as specified in Section 5.3.2.4.4, VVoIP NMS Interface Requirements. | 5.3.2.8.3.1 | Y | | |
| 314 | As specified in Section 5.3.2.4.4, VoIP NMS Interface Requirements, the MFSS, WAN SS, and LSC components shall support at least one pair of physical Ethernet management interfaces at the component | 5.3.2.17.2 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| | level (not at the device level). One of these Ethernet management interfaces shall be used for component-level communication with a Local EMS. The other Ethernet management interface shall be used for component-level communication with the remote VVoIP EMS. | | | | |
| 315 | A network appliance shall have Operations interfaces that provide a standard means by which management systems can directly or indirectly communicate with and, thus, manage the various network appliances in the DISN. | 5.3.2.17.2 | Y | Y | Y |
| 316 | There shall be a local craftsperson interface (Craft Input Device (CID)) for OA&M for all VVoIP network components. | 5.3.2.17.2 | Y | Y | Y |
| 317 | The network appliances shall provide NM data to the external VVoIP EMS. | 5.3.2.17.2 | Y | Y | Y |
| 318 | A network appliance shall communicate with an external Voice and Video management system by a well-defined, standards-based management interface using an industry-accepted management protocol. | 5.3.2.17.2 | Y | Y | Y |
| 319 | Communications between VVoIP EMS and the VVoIP network appliances shall be via IP. | 5.3.2.17.2 | Y | Y | Y |
| 320 | A network appliance shall issue state change notifications for changes in the states of replaceable components, including changes in operational state or service status, and detection of new components. | 5.3.2.17.2 | Y | Y | Y |
| 321 | A network appliance shall be provisioned by the VVoIP EMS with the address, software, and OSI Layer 4 port information associated with its Core Network interfaces. | 5.3.2.17.2 | Y | Y | Y |
| 322 | A network appliance shall be capable of maintaining and responding to VVoIP EMS requests for resource inventory, configuration, and status information concerning Core Network interface resources (e.g., IP or MAC addresses) that have been installed and placed into service. | 5.3.2.17.2 | Y | Y | Y |
| 323 | Network appliances that provide voice and video call service shall have the capability to invoke traffic flow (NM) controls as detailed in Section 5.3.2.18, Network Management Requirements of Appliance Functions. | 5.3.2.17.2 | Y | Y | Y |
| 324 | A network appliance shall be capable of setting the Administrative state and maintaining the Operational state of each Core Network interface, and maintaining the time of the last state change. | 5.3.2.17.2 | Y | Y | Y |
| 325 | Alarm messages must be distinguishable from administrative log messages. | 5.3.2.17.3.1.1 | Y | Y | Y |
| 326 | The NEs shall detect their own fault (alarm) conditions. | 5.3.2.17.3.1.2 | Y | Y | Y |
| 327 | The NEs shall generate alarm notifications. | 5.3.2.17.3.1.3 | Y | Y | Y |
| 328 | The network elements shall send the alarm messages in NRT. More than 99.95 percent of alarms shall be detected and reported in NRT. Near Real Time is defined as event detection and alarm reporting within 5 seconds of the event, excluding transport time. | 5.3.2.17.3.1.4 | Y | Y | Y |
| 329 | The network components shall send alarm messages in SNMPv3 format. | 5.3.2.17.3.1.5 | Y | Y | Y |
| 330 | Capability to access and modify configuration data by the VVoIP EMS shall be controllable by using an access privileges function within the network appliance. | 5.3.2.17.3.2.1 | Y | Y | Y |
| 331 | The VVoIP NEs shall be able to receive and respond to remote NM commands. | 5.3.2.17.3.4.2 | Y | Y | Y |
| 332 | When ASAC budgets are reduced, by NM action, below the current budget allocation, any previous sessions (regardless of precedence level) in excess of the new budget shall be allowed to terminate naturally. This assumes that the CE Router queue bandwidths would not be reduced until the LSC session count fell below or equal to the newly commanded reduced budget, to prevent the corruption of existing sessions. | 5.3.2.17.3.4.2.2 | Y | Y | Y |
| 333 | The LSC, MFSS, and WAN SS shall have the capability of setting the percentage of calls to be blocked to the designated destination(s). | 5.3.2.17.3.4.2.7 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 334 | FLASH and FLASH-OVERRIDE calls shall not be affected by NM controls. | 5.3.2.17.3.4.2.7 | Y | Y | Y |
| 335 | Within IP, directionalization is controlled by designating all or part of the call budget as inbound (i.e., local destination) and/or outbound (i.e., local origination). The default is no designation (i.e., calls up to the total budget can be inbound or outbound in any combination). It does not change the total budget, only the sourcing direction of the budget; therefore, there is no impact to the router queue bandwidths. | 5.3.2.17.3.4.2.10 | Y | Y | Y |
| 336 | Within IP, the routing of all traffic (i.e., VVoIP and non-VVoIP) is handled via MPLS in the DISN core. The MPLS automatically finds the most effective route for the traffic. | 5.3.2.17.3.4.2.11 | Y | Y | Y |
| 337 | The WAN-level ASAC must be able to account for each subtended LSC under its control. Therefore, the MFSS and WAN SS ASAC must be able to set call budgets for multiple LSC locations via the VVoIP EMS and local EMS access points. | 5.3.2.17.3.4.2.13 | Y | | Y |
| 338 | The LSC-level ASAC is required to only account for itself. Therefore, the LSC ASAC must be able to set call budgets for only the PEI/AEIs under its control via the VVoIP EMS and local EMS access points. | 5.3.2.17.3.4.2.13 | | Y | |
| 339 | The product shall have the capability of setting a PEI/AEI's maximum allowed precedence level for originating a call. This is a "subscriber class mark feature," which is controlled by the LSC system administrator. | 5.3.2.17.3.4.2.14 | | Y | |
| 340 | The product shall have the capability of controlling the destination(s) that a PEI or AEI is restricted from calling. This is a subscriber class mark feature that is controlled by the LSC system administrator. | 5.3.2.17.3.4.2.14 | | Y | |
| 341 | The ASAC must provide the separate counts for voice and video, in 5-minute intervals. The MFSS and WAN SS ASAC must provide these counts for each of the subtended LSCs under its control, while the LSC is only to provide these counts for the PEIs/AEIs that it controls. | 5.3.2.18.2 | Y | Y | Y |
| 342 | A switching network appliance shall acquire, activate, and manage a CCA software download as directed by the Local EMS. The CCA software may be managed on a per CCA hardware component basis. | 5.3.2.18.3.1.1 | Y | Y | Y |
| 343 | The CCA shall be able to manage the following parameters in the CCA from the VVoIP EMS:<br>• CCA Identification parameter<br>• Recording Office Identification parameter | 5.3.2.18.3.1.1 | Y | Y | Y |
| 344 | The CCA shall manage the activation and deactivation of service features. The CCA shall maintain data for the media server and UFS functions it interacts with. The CCA shall be able to create a backup and manage restoration of configuration data by placing its stable data and changes to the latest configuration in a nonvolatile storage device. | 5.3.2.18.3.2 | Y | Y | Y |
| 345 | A CCA shall meet all applicable Operations Technology Generic Requirements (OTGR) for switching system NE trouble isolation in Telcordia Technologies GR-474-CORE. A CCA shall perform root-cause analysis for any faults within its purview, report the root cause, and suppress the reporting of non-root-cause conditions. A CCA shall support the ability to perform internal diagnostics on its call processing functionality and internal resources, initiated either locally or upon request by the VVoIP EMS. | 5.3.2.18.3.3 | Y | Y | Y |
| 346 | The CCA shall provide trunk group-related traffic measurements as specified in Telcordia Technologies GR-477-CORE, Section 4.1.3. For all calls originating at a CCA, the CCA shall monitor call set-up delay statistics, including delay incurred as part of the set-up of the core network bearer connection. | 5.3.2.18.3.5 | Y | Y | Y |
| 347 | An MG shall manage logical and physical resource inventory information. An MG shall issue an autonomous notification to the VVoIP EMS whenever a new inventory or capabilities are added, or configuration is changed through local management activity. An MG maintains the information related to service features and data, including the management of service logic. | 5.3.2.18.5.1 | Y | Y | Y |
| 348 | An MG shall manage current MG state and status information about its installed major components, line and plug-in cards, and processes. | 5.3.2.18.5.1.2 | Y | Y | Y |
| 349 | Upon the detection or clearing of alarm conditions, the MG shall generate and forward, based on filtering criteria, a notification to the VVoIP EMS. An MG shall support queries for alarm status, state, and current problem information. An MG shall monitor, detect, and | 5.3.2.18.5.2.1 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| | generate alarm conditions and states associated with hardware, functional components, system interfaces, and logical resources (e.g., trunk terminations, tone and announcement generators, media content detectors, signal processors, echo control devices). | | | | |
| 350 | An MG shall perform root-cause analysis for any faults within its purview, report the root cause, and suppress the reporting of non-root-cause conditions. | 5.3.2.18.5.2.2 | Y | Y | Y |
| 351 | An MG shall, on request or per a pre-established schedule, run diagnostics on internal resources, hardware, or software, and report the result to the VVoIP EMS. | 5.3.2.18.5.2.3 | Y | Y | Y |
| 352 | An MG shall provide both local and remote loopback capabilities for the digital interfaces that terminate at the MG ports. | 5.3.2.18.5.2.3 | Y | Y | Y |
| 353 | Upon receiving a request from the VVoIP EMS or by an established schedule, an MG shall provide a report of a parameter's present or history counters. | 5.3.2.18.5.3.1 | Y | Y | Y |
| 354 | An MG shall generate TCAs to notify the VVoIP EMS when a thresholded count exceeds its threshold during a measurement interval. | 5.3.2.18.5.3.2 | Y | Y | Y |
| 355 | The MG shall manage interexchange trunk (between MG and SSP), trunk group, trunk, and physical resource inventory and configuration data. The MG shall manage MG termination-related status information. | 5.3.2.18.5.4 | Y | Y | Y |
| 356 | An MG shall, on request or on schedule, run diagnostics on internal resources and hardware, run checks on software, and report the results to the VVoIP EMS. The MG shall provide test access to external test equipment for passively monitoring the traffic through the MG interfaces. This passive monitoring shall not degrade the performance of traffic. | 5.3.2.18.5.5 | Y | Y | Y |
| 357 | The MG shall receive voice-grade analog line configuration data from the VVoIP EMS upon service activation. | 5.3.2.18.5.7 | Y | Y | Y |
| 358 | The MG shall provide diagnostic tests to detect and verify faults, such as low loop resistance or ground conditions, or any other faults within the MG that could cause false ring trip or false answer. | 5.3.2.18.5.8 | Y | Y | Y |
| 359 | The MG shall support the collection of the standard DS1, DS3, Physical Layer Convergence Protocol (PLCP), SONET, and ISDN BRI line performance monitoring requirements, as defined in Telcordia Technologies GR-820-CORE, for applicable interfaces. | 5.3.2.18.5.9 | Y | Y | Y |
| 360 | For the selected recording format that is chosen, of all the call information that will be provided, the following call data shall be provided in the record data:<br>1. Host Name of the CCA controlling the call processing.<br>2. Start Date of call (In Julian or Calendar).<br>3. Start Time of Call (Hour + Minute + Second).<br>4. Elapsed Time of Call and/or Stop Time of call.<br>5. Calling Number.<br>6. Called Number (included all dialed digits). | 5.3.2.19.2.1 | Y | Y | Y |
| 361 | For the selected recording format that is chosen, of all the call information that will be provided, the following call data shall be provided in the record data if it applies to the call: Conference Call Indicator. | 5.3.2.19.2.1 | Y | Y | Y |
| 362 | The product shall provide a voice quality record at the completion of each voice session. The voice quality record shall be included in the CDR that the LSC, MFSS, or WAN SS generates for that session, and shall conform to the E-Model, as described in TIA TSB-116-A, and ITU-T Recommendation G.107. The voice quality record shall contain the calculated R-Factor for the Voice session per TIA TSB-116-A. | 5.3.2.19.2.1.1 | Y | Y | Y |
| 363 | As part of the voice quality record, the product shall provide the raw voice session statistics that are used to make the R-Factor calculation to include, as a minimum, the latency, packet loss, Equipment Impairment Factor (Ie), and the TCLw. The product shall provide the jitter for the session. | 5.3.2.19.2.1.1 | Y | Y | Y |
| 364 | The product shall generate an alarm to the VVoIP EMS when the session R-Factor calculation in the CDR fails to meet a configurable threshold. By default, the threshold shall be an R-Factor value of 80, which is equivalent to an MOS value of 4.0. | 5.3.2.19.2.1.1 | Y | Y | Y |

## Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|
| 365 | The mass storage in the BA must be non-volatile. The mass storage in the BA must be able to retain at least five average-busy-season business days of AMA data. (NOTE: This is needed to provide adequate capacity for high-volume storage of CDRs.) | 5.3.2.19.2.3 | Y | Y | Y |
| 366 | The BA should be able to output the records electronically over a secured connection. The BA should have the ability to transfer the records to a physical storage media that is also removable. | 5.3.2.19.2.4 | Y | Y | Y |

**LEGEND:**

ACD Automatic Call Distributor
AEI AS-SIP End Instrument
AMA Automatic Message Accounting
ANAT Alternative Network Address Types
ANSI American National Standards Institue
ASAC Assured Services Admission Control
ASLAN Assured Services Local Area Network
AS-SIP Assured Services Session Initiation Protocol
ATQA Attendant Queue Announcement
B2BUA Back-to-back User Agent
BA Billing Agent
BER Bit Error Rate
BLV Busy Line Verification
BNF Backus-Naur Form
C Conditional
C2 Command and Control
CAC Common Access Card
CAS Channel Associated Signaling
CCA Call Connection Agent
CCS7 Common Channel Signaling 7
CDR Call Data Record
CE Customer Edge
CF Call Forward
CH1 Change 1
CID Craft Input Device
CND Calling Number Delivery
CONUS Continental United States
D-Channel Data Channel
DB Database
DHCP Dynamic Host Configuration Protocol
DISA Defense Information Systems Agency
DISN Defense Information System Network
DN Directory Number
DoD Department of Defense
DS1 Digital Signal Level 1
DS3 Digital Signal Level 3
DSCP Differentiated Services Code Point
DSN Defense Switched Network
E2E End-to-end
EBC Edge Boundary Controller
EC Echo Canceller
EI End Instrument
EMS Element Management System
EO End Office
ETSI European Telecommunications Standards Institute
FCAPS Fault, Configuration, Accounting, Performance, and Security
FoIP Fax over Internet Protocol
FQDN Fully Qualified Domain Name
G3 Fax Group 3 Facsimile
Hz Hertz
IAD Integrated Access Device
IAW In Accordance With
ICA Isolated Code Announcement
ID Identification
ICMPv6 Internet Control Message Protocol for IPv6
Ie Equipment Impairment Factor
IEEE Institute of Electrical and Electronics Engineers, Inc.
IETF Internet Engineering Task Force
IP Internet Protocol
IPB IP ASAC Budget
IPC IP ASAC Call Count
IPSec Internet Protocol Security
IPv4 Internet Protocol Version 4
IPv6 Internet Protocol Version 6
kbps Kilobytes per second
LAN Local Area Network
LDAP Lightweight Directory Access Protocol
LDAPv3 Lightweight Directory Access Protocol version 3
LSC Local Session Controller
MAC Media Access Control
Mbps Megabytes per second
MFS Multifunction Switch
MFSS Multifunction SoftSwitch
MG Media Gateway
MGC Media Gateway Controller
MLD Multicast Listener Discovery
MLPP Multilevel Precedence and Preemption
Modem Modulator/Demodulator
MoIP Modem over Internet Protocol
MOS Mean Opinion Score
MPLS Multiprotocol Label Switching
ms milliseconds
MTU Maximum Transmission Unit
NE Network Element
NM Network Management
NMS Network Management System
OA&M Operations, Administration, and Maintenance
OCONUS Outside the Continental United States
OSI Open Systems Interconnect
OTGR Operations Technology Generic Requirements
PBAS Precedence Based Assured Services
PBX Private Branch Exchange
PCD Precedence Call Diversion
PCM Pulse Code Modulation
PEI Proprietary End Instrument
PLCP Physical Layer Convergence Protocol
PRI Primary Rate Interface
PSTN Public Switch Telephone Network
REL Release Message
RFC Request For Communication
RTS Real Time Services
SAC Session Admission Control
SBU Sensitive, but Unclassified
SCIP Secure Communications Interoperability Protocol
SCS Session Control and Signaling
SDP Session Description Protocol
SG Signaling Gateway
SIP Session Initiation Protocol
SMEO Small End Office
SONET Synchronous Optical Network
SRTCP Secure Real-Time Transport Control Protocol
SRTP Secure Real-Time Transport Protocol
SS Softswitch
SS7 Signaling System number 7
SUT System Under Test
TA Terminal Adaptor
TCA Threshold Crossing Alert
TCLw Weighted Terminal Coupling Loss
TDM Time Division Multiplexing
TIA Telecommunications Industry Association
TLS Transport Layer Security
UAC User Agent Client
UAS User Agent Server
UC Unified Capabilities
UCR Unified Capabilities Requirements
UDP User Datagram Protocol
URI Uniform Resource Identifier
U.S. United States

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

| ID | Requirement | UCR Ref (UCR 2008 CH1) | MFSS | LSC | WAN SS |
|---|---|---|---|---|---|

| | | | |
|---|---|---|---|
| ISDN | Integrated Services Digital Network | VBD | Voice Band Data |
| ISUP | ISN User Part | VoIP | Voice over Internet Protocol |
| ITU-T | International Telecommunications Union – Telecommunication Standardization Sector | VVoIP | Voice and Video over Internet Protocol |
| | | WAN | Wide Area Network |
| IWF | Interworking Function | Y | Yes |